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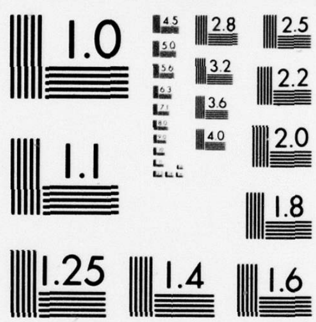
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(6) REVERBERATION TIME MEASUREMENTS IN AN UNDERWATER CHAMBER WITH AND WITHOUT A PRESSURE-RELEASE WALL COVERING.

(10) Richard J. Fridrich

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The Pennsylvania State University

The Graduate School

Graduate Program in Acoustics

Reverberation Time Measurements in an Underwater Chamber With and Without A Pressure-Release Wall Covering

A Thesis in

Acoustics

by

Richard J. Fridrich

Submitted in Partial Fulfillment
of the Requirements
for the Degree of

Master of Science

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We approve the thesis of Richard J. Fridrich.

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TABLE OF CONTENTS

LIST OF TABLES

LIST OF FIGURES

LIST OF FIGURES (Continued)

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CHAPTER I

BACKGROUND AND INTRODUCTION

1.1 Background

The practice of noise control engineering often involves the measurement of acoustical quantities in order to determine the effect of modifications to a mechanical system on the noise radiated by that system. Of the many different quantities that could be used to describe the effectiveness of the modifications, an important one is the sound power radiated by the system. Knowing the sound power output of a machine and the environment in which it is operating, one can predict the values of many other acoustical quantities when measured in the far field. It is this prediction capability that makes sound power such a useful quantity in noise control.

There are several methods for measuring the sound power output of a machine, each with its own advantages and disadvantages. Certain methods require free-field conditions, such as in an anechoic room, in order to determine the directivity pattern of the radiated sound. Directivity patterns, however, are not always needed, and since the methods that produce them are long and difficult, simpler methods using reverberation rooms are generally used. The two methods of determining the sound power output of a machine in a reverberant room are the direct and comparison methods which are described in detail in American National Standard (ANS) S1.21-1972 Methods for the Determination of Sound Power Levels of Small Sources in Reverberation Rooms.

The comparison method determines the sound power output of a machine by comparing its sound pressure level in the reverberant field with the sound pressure level generated by a reference source of known sound power. This method, of course, can only be used if a known reference source is available. The direct method arrives at the sound power output by using the measured sound pressure level in the reverberant field along with a knowledge of the absorption present in the room which is obtained from measurements of the reverberation time and the total surface area and volume of the room. The procedure for measuring the reverberation time is described in ANS S1.7-1970 (ASTM C 423-66) (R1972) Standard Method of Test for Sound Absorption of Acoustical Materials in Reverberation Rooms. Reverberation time is defined as the time it takes for the sound field in a room to decay 60 decibels after the source of the sound has been shut off. Briefly, the measurement procedure calls for exciting the room with sound from a source, abruptly shutting off the sound at the source, and recording the response of a receiver during the decay of the sound field in the room. If a full 60-decibel delay is not obtained, the reverberation time is obtained by linear extrapolation using the slope of the recorded decay.

The methods which have been briefly summarized above have long been used for determining the sound power output of machines which operate in air. There are as yet no standards on sound power output which deal with water as the medium of propagation. The usefulness of the standards of air acoustics in situations involving water as a medium was investigated by Blake and Maga.[1] Their results indicate

that the standards for sound power measurements in reverberation rooms are as valid in water as in air. However, certain important differences between water and air affect the degree of adherence to the standards which can be achieved in underwater reverberation rooms.

Since the speed of sound is about five times greater in water than in air, the wavelength of a sound of a given frequency is about five times larger in water than in air. This implies that, with respect to wavelength, a room of a given size appears to be about five times smaller in water than in air. Because of this, the requirement in the standards that the locations of the measurement hydrophones/ microphones be at least a half wavelength from the room surfaces and from each other is much more restrictive in water than in air. In order to meet this requirement, each dimension of a room in water would have to be five times longer than the corresponding dimensions of a room which fulfills the standards in air. Another important consideration is the difference in the characteristic impedance of water versus air. Because the impedance of water is so much larger than air, the absorption of sound is greater for most materials in water than in air. This is because most materials used in construction have characteristic impedances which more closely match the impedance of water than that of air. As a result, an air-water surface reflects sound better, for example, than a concrete-water surface. Ultimately, this means that a room of a given size and construction is more reverberant in air than in water.

1.2 Introduction

The Machinery Quieting Laboratory (MQL) in the Applied Research Laboratory at The Pennsylvania State University is a facility devoted to noise control in underwater machines. Noise reduction programs on underwater machines have been carried out at this facility by measuring the sound pressure levels generated by the machinery before and after noise reduction modifications, thereby obtaining an indication of the relative effectiveness of the modifications. The noise reduction measured by this method is not an absolute measure but is rather a comparative one which may depend upon the environment in which the measurements are made. The validity of this method rests on the assumption that no changes in the test environment have occurred during the course of the measurements.

This method of noise control, while workable, has the limitation that, even though the noise from the machine may be reduced under laboratory conditions, it is not possible to determine with any certainty the sound radiated under field conditions. This limitation need not exist, however, if the sound power output of the machine is measured and used to determine the effectiveness of the noise control modifications. As discussed earlier, the sound power output of a machine can be measured by several different methods. However, the nature of the available facilities often dictates which methods can be used. This is true in the case of the Machinery Quieting Laboratory since the methods which require an anechoic chamber cannot be used because the available test chamber is known to have characteristics somewhere in between an anechoic and a reverberant chamber. Since the

MQL test chamber is too reverberant to permit the use of the methods requiring an anechoic chamber, the only methods available are those which require a reverberant room. Before either the comparison or direct methods can be used, however, certain questions must be answered in order to insure that, for the method chosen, the desired result is obtained.

The standard on determining sound power using a reverberant room assumes that the source being tested is small. In the case of the Machinery Quieting Laboratory, however, this assumption often cannot be made since the dimensions of the test machines are frequently a significant percentage of the dimensions of the test chamber. Because of the size of the test machines, the comparison method encounters difficulties in locating a suitable position for the reference source when it is substituted in place of the test machine. Because the reference source is small, the discrepancy in size between the reference source and the test machine causes uncertainty in any results obtained from arbitrarily choosing a location for the reference source. Since it is possible that the test machine might not act like a point source, but rather like a line source or some other kind of extended source, finding a unique location for the reference source may be impossible. The use of the comparison method is highly questionable under these circumstances. To answer the questions of what type of source the test machine is and where the reference source should be located requires anechoic conditions which are not available in the Machinery Quieting Laboratory.

The only remaining method for determining sound power is the direct method. Once again, the large dimensions of the test machine prevent strict adherence to the standard, but at least it is not necessary to determine what kind of source the test machine is in order to use this method as was the case with the comparison method. The quantities which must be determined in the direct method are the average sound pressure level in the reverberant field and the absorption present in the test chamber. The measurement of the average sound pressure level in the MQL test chamber can be accomplished without difficulty using the measuring system of the comparative noise reduction programs. The absorption characteristics of the MQL test chamber, however, are not well known and must be determined from reverberation time measurements. As with the measurement of sound power levels under water, the measurement of reverberation time in underwater chambers is not a routinely performed measurement and, as a result, no standard exists which specifically deals with underwater reverberation measurements.

Two preliminary measurements of the reverberation time in the MQL tank have been performed by D. S. Pallett and by D. W. Ricker for several widely spaced frequency bands. Their results were not detailed enough for use in sound power measurements, but they did indicate that the absorption in the MQL tank was rather high for a good reverberation room. A possible method for lowering the absorption by covering the walls of the MQL tank with a pressure-release material was suggested by Pallett. This thesis implements that suggestion. Because there is so little published literature on the measurement of reverberation time

in underwater chambers, extensive decay time measurements were made under various conditions in order to obtain additional information on the possible effects on the reverberation time of the placement of the source, the placement of the receiver, and the presence of the test machine in the tank.

1.3 Thesis Objective

The primary objective of this thesis is to determine by reverberation time measurements the absorption characteristics of the MQL test chamber with and without a pressure-release wall covering. The effectiveness of the pressure-release wall covering in reducing the absorption in the chamber is demonstrated by an increase in the reverberation time measured with the wall covering present as compared to the reverberation time measured without the wall covering present. The detailed measurement of the reverberation time of the MQL test chamber provides sufficient information on the acoustical environment to permit the possible use of the direct method for determining the sound power levels of machines tested in the Machinery Quieting Laboratory.

CHAPTER II

EXPERIMENTAL APPARATUS AND METHODS

2.1 MQL Tank

The MQL tank, as shown in Figures 1 and 2, is a rectangular tank of dimensions 7.62 meters by 2.44 meters by 2.44 meters. The walls and floor of the tank are constructed of 0.3048 meter thick concrete, and the top is completely open. An air space exists behind 50% of the tank walls, a water storage tank behind 17%, and earth fill behind the remaining 33% of the walls and underneath all of the tank floor. Within the tank are three adjustable clamping supports for underwater machinery. In addition, the tank has provisions for water-tight coupling of the test machinery to a dynamometer located outside of the tank. When the tank is filled to a depth of 2.13 meters, the water-filled volume is approximately 39.64 cubic meters. The total surface area of this volume of water is approximately 80.1 square meters. Approximately 23.2% of this area is the air-water boundary of the open top of the tank, while the remaining 76.8% is the water-concrete boundary formed by the floor and walls of the tank.

2.2 Pressure-Release Wall Covering

The pressure-release wall covering used in this experiment was Rubatex G-231-N, a closed-cell neoprene rubber available in 6.35 millimeter thick sheets approximately 1 meter wide by 3.048 meters long. Since permanent installation of the neoprene wall covering

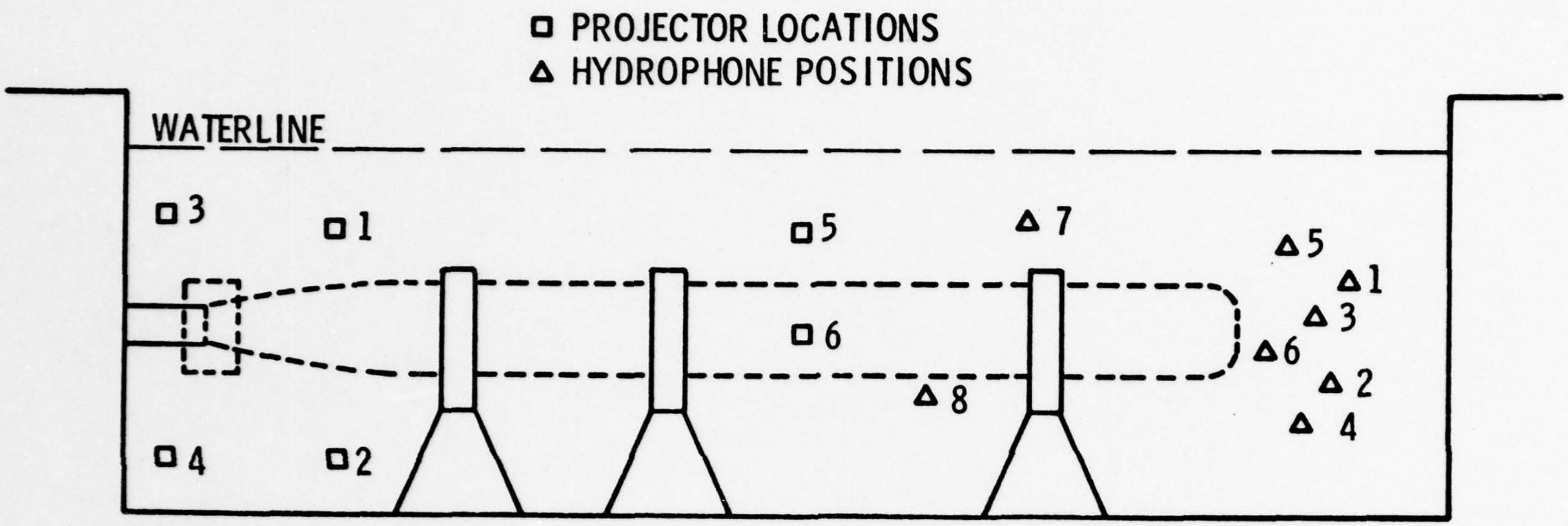

Figure 1. MQL Tank - Side view showing projector locations and hydrophone positions.

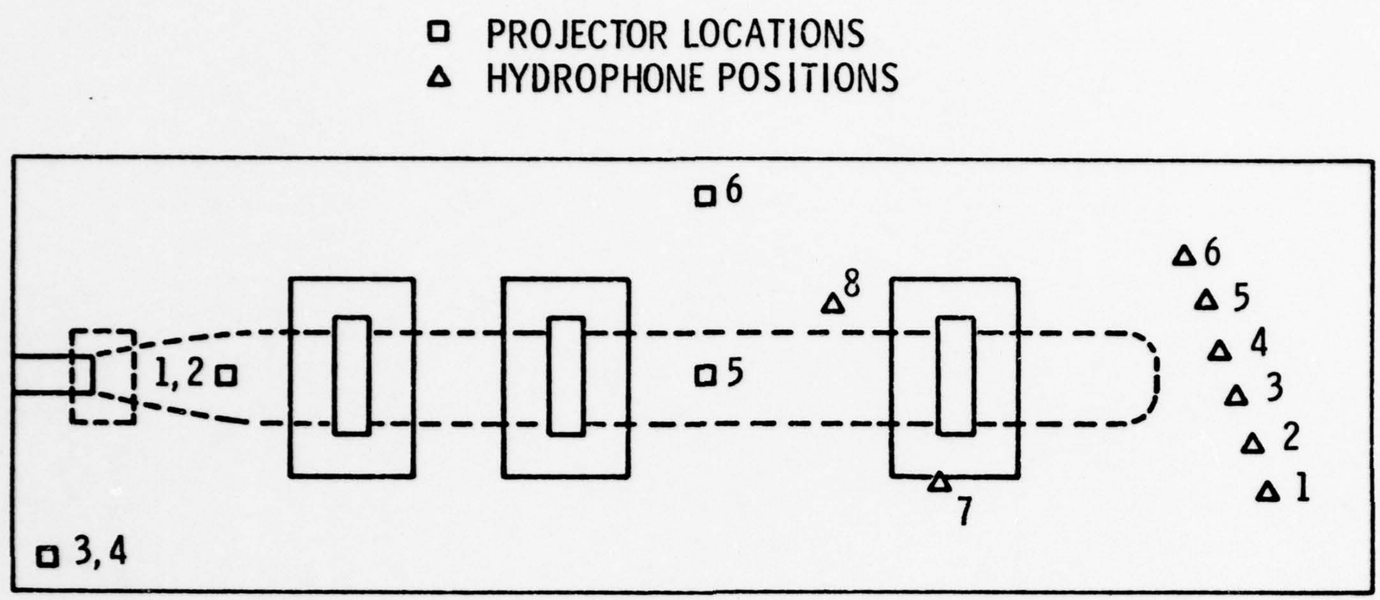

Figure 2. MQL Tank - Top view showing projector locations and hydrophone positions.

would drastically reduce the flexibility of the MQL test facilities, a set of close-fitting, removable curtains was constructed from the 6.35 millimeter neoprene sheeting which could be lowered into the tank as desired. To counteract the buoyancy of the closed-cell neoprene, steel weights were attached about 5 centimeters from the bottom of the neoprene sheet. The steel weights consisted of two steel bars with dimensions 96.52 centimeters by 8.25 centimeters by 1.27 centimeters, and a total mass of 15.34 kilograms. Attaching the steel weights to the neoprene was accomplished by bolting the two steel bars together with the neoprene sandwiched in between as shown in Figure 3.

At the top of the curtain, the neoprene sheet was wrapped around a 95 centimeter long section of 2.54 centimeter ID thin-wall conduit. To secure the conduit in position, two aluminum bars with dimensions 95 centimeters by 1.9 centimeters by 0.327 centimeters were bolted together with a double thickness of neoprene sandwiched in between as illustrated in Figure 3. The centerline of the conduit was positioned on the unstretched neoprene sheet 2.31 meters above the bottom of the steel weights so that when the finished curtain was installed in the tank, a 12.7 centimeter stretch was introduced into the neoprene. It was found that this amount of stretch kept the neoprene curtain taut when the tank was filled with water, thus correcting for the stretch caused by the hydrostatic forces on the neoprene.

When the curtains were installed in the tank, the steel weights rested on the bottom, while the tops of the curtains were held in position by long sections of 1.27 centimeter ID galvanized pipe which were passed through the conduit at the top of each curtain. The

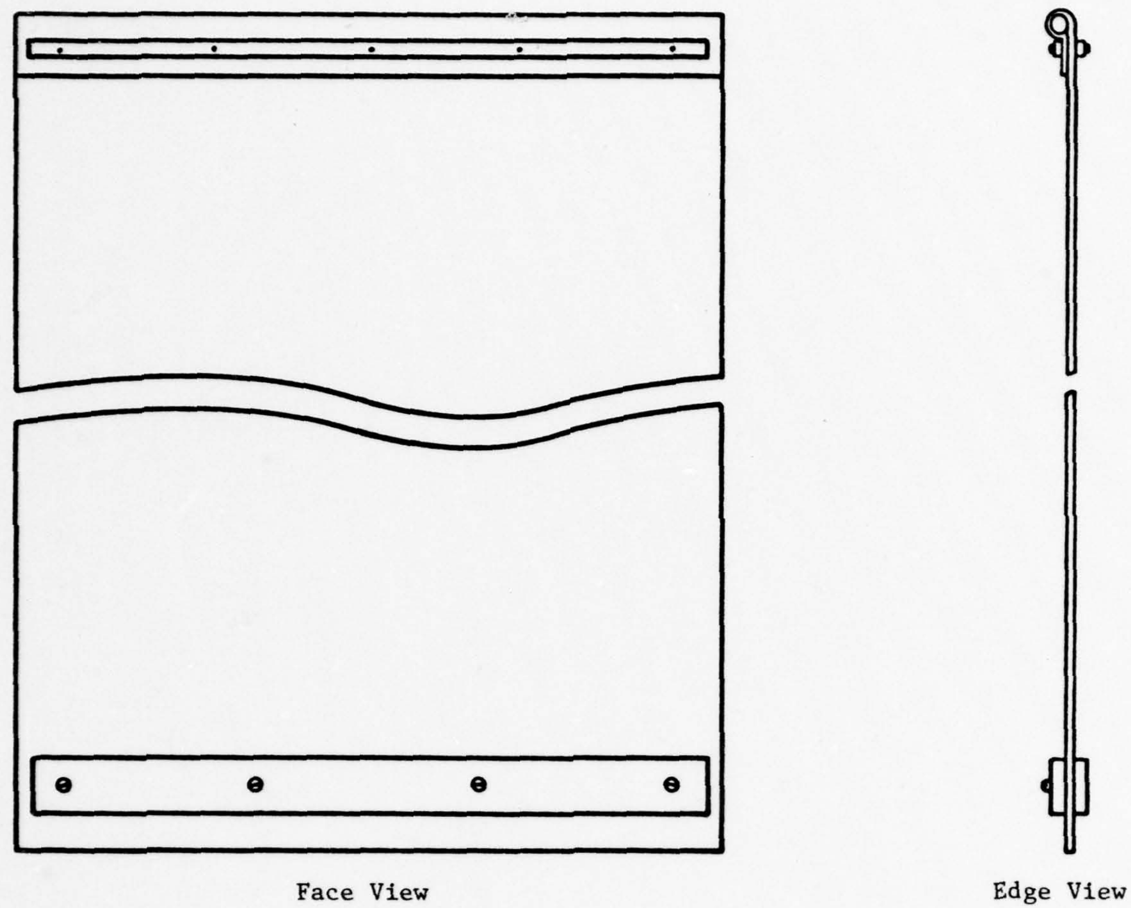

Figure 3. Neoprene curtain construction.

curtains were positioned as close to the walls as possible by keeping both the steel weights and the tops of the curtains in contact with the wall surfaces. The completed set of neoprene curtains covered the wall surfaces of the MQL tank with one thickness of neoprene. Total estimated coverage by the neoprene curtains was 42.92 square meters or about 53.6% of the total surface area of the tank.

2.3 Source Instrumentation

As described earlier, the first step in reverberation time measurements is to excite the room using a sound source. A block diagram of the instrumentation used in the excitation of the MQL tank is presented in Figure 4. The sound source in this experiment was a Massa Projector (Model TR-97). The excitation signal was obtained by filtering the output of a General Radio Random Noise Generator (Type 1390-B) into the desired one-third octave band using a General Radio Sound and Vibration Analyzer (Type 1564-A). The resulting signal was amplified by a CML Power Amplifier (Model A2KM6304) which drove the Massa projector. A simple toggle switch, located between the one-third octave filter and the power amplifier, was used to abruptly interrupt the signal as required by the measurement procedure. The output of the CML amplifier was monitored using a Tektronix Oscilloscope (Type 321A) with a 1000-to-1 Probe (Type P6013A) in conjunction with a B & K Sound Level Meter (Type 2209).

In order to assess the effect of source placement on reverberation time, six projector locations were used in this experiment as shown in Figures 1 and 2. Projector locations one and two were chosen so as to simulate the normal location of noise generating machinery

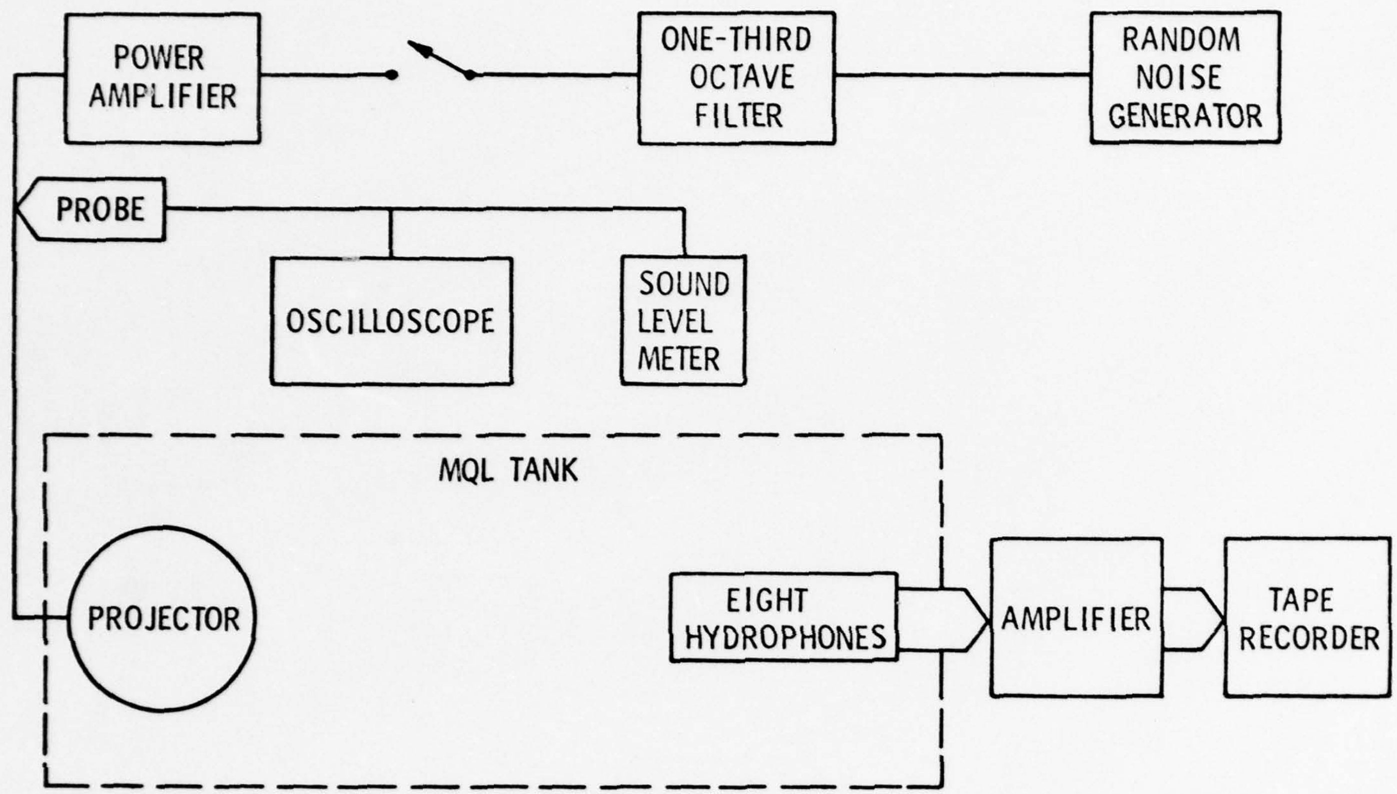

Figure 4. Block diagram of source and receiver instrumentation.

the MQL tank was designed to test. Projector locations three and four were chosen using the recommendations for source locations given in ANS S1.7-1970 (ASTM C423-66) (R1972) Method of Test for Sound Absorption of Acoustical Materials for Reverberation Rooms. This standard recommends that the sound source be located "in the trihedral corners of the room." The choice of these locations is based on a desire to excite the oblique modes of the room more strongly than would otherwise be possible using other choices of projector locations. Projector locations five and six were chosen for geometric as well as acoustical reasons. By locating a source at location five, which represents the geometric center of the tank, the axial modes of the tank are more strongly excited than the other modes. Projector location six, which retains much of the symmetry of location five, was chosen for comparison with projector location five. If locating a projector near a major surface area of the tank causes the reverberation time of the tank to be affected, that effect should be discernible by comparing the reverberation times obtained with the projector at location five with those obtained with the projector at location six. Likewise, other effects on the reverberation time due to source placement should become apparent through a comparison of the results obtained from these six projector locations.

2.4 Receiver Instrumentation

Once the test room has been excited by a sound source, the decay of the sound field after the excitation signal has been shut off was recorded. Figure 4 is a block diagram of the instrumentation used to receive and record the decay of the sound field in the MQL tank. In

this experiment, the sound field in the MQL tank was monitored by eight Bendix Hydrophones (Model DX 579) located in the tank. The signals from the hydrophones were amplified and recorded on 2.54 centimeter wide Ampex Magnetic Tape (Type 76657G111) using a fourteen-channel Sangamo Tape Recorder (Model 4742) operated at a speed of 38 centimeters per second.

The hydrophones used in this experiment were located at eight fixed positions as shown in Figures 1 and 2. Hydrophone locations one through six were hydrophone locations which had been used in previous noise reduction programs and were used again in this experiment to provide some continuity with those programs. Hydrophone locations seven and eight were chosen arbitrarily, but with a degree of caution so that the selected locations would not be too close to any major surface area of the tank, to any of the other hydrophones, or to the region of space which might be occupied by potential test machinery. The hydrophones were suspended by their own cables from two angle-iron bars which spanned the open top of the tank. Attached to each hydrophone to minimize undesired motion was a small cylindrical brass weight which was threaded onto the hydrophone's shielded cable and allowed to rest on top of the waterproofing which surrounds the hydrophone element.

2.5 Test Conditions

The reverberation time measurements for this experiment were performed under four different test conditions and recorded for later analysis. Reverberation measurements performed in the water-filled MQL tank without the neoprene curtains installed were designated as

test condition one. Measurements under test condition two were also made without the neoprene curtains present, but included a MK48 torpedo as representative of a machine undergoing noise reduction tests in the MQL tank. The torpedo was not operated during this experiment, but was included in the tank to determine its effect on the reverberation time as a large stationary diffuser. Measurements under test condition three were made with the neoprene curtains installed, but without the torpedo present. The final set of measurements, designated as test condition four, were made with both the neoprene curtains and the torpedo in place. The effect on the reverberation time due to the presence of the neoprene curtains can be evaluated by comparing the results of either test conditions one and three or two and four. The effect on the reverberation time due to the introduction of a large stationary diffuser can be evaluated by comparing the results of either test conditions one and two or three and four.

For each of the abovementioned test conditions, two other parameters were varied during this experiment. The first parameter was the placement of the sound source used to excite the room which was discussed earlier. The second parameter was the center frequency of the one-third octave band into which the excitation signal was filtered before being fed to the projector.

In this experiment, the center frequency of the filter was varied so as to cover a range of four octaves from 1 kilohertz to 16 kilohertz. This was done by setting the center frequency of the filter to thirteen different frequencies from 1 kilohertz to 16 kilohertz in accordance with the standard preferred center frequencies specified in ANS

S1.11-1966 (R1975) Specification for Octave, Half-Octave, and Third-Octave Band Filter Sets. By varying these two parameters for each test condition, the spatial and spectral characteristics of the reverberation time in the MQL tank were investigated.

2.6 Data Taking Procedure

The procedure for recording the reverberation data was as follows. Upon the completion of the preparations for each test condition, the MQL tank was filled with water to a depth of about 2.13 meters. To allow the tank to reach steady-state conditions, 48 hours were allowed to elapse between the time the tank was filled and the first measurement recorded. This length of time is sufficient for the air bubbles generated by the filling process to dissipate enough so that the day-to-day variation of the measured reverberation time is within 10% of the steady-state value.[1] Blake and Maga attributed the variation of the reverberation time during the time immediately following the filling of the tank to resonance effects in the bubbles which increase the sound absorption in the water.

After the waiting period was over, the projector was positioned in one of the six source locations, and the excitation signal was turned on. The desired one-third octave band was selected on the General Radio sound and vibration analyzer, and the signal input to the projector was adjusted by increasing the gain of the CML amplifier while monitoring the Tektronix oscilloscope and the B & K sound level meter. Next, the amplifier gain for each hydrophone was adjusted for proper record level. Upon completing all the adjustments, the input signal voltage to the projector was documented along with the amplifier

gain settings for each hydrophone. The excitation signal was momentarily shut off while the tape recorder was started. Once the tape recorder was up to speed, the excitation signal was again turned on, and the documented voltages and gain settings were verified. The response of the hydrophones to the excitation signal was recorded for approximately 10 seconds before the excitation signal was shut off using the toggle switch. To insure that the decay of the sound field had been recorded, the tape recorder was permitted to continue recording for at least 10 seconds after the excitation signal had been shut off. The total recording time for each measurement was, therefore, at least 20 seconds. Each measurement was vocally documented using an independent voice track on the recorder.

Using this procedure, 26 measurements were made for each projector location of each test condition. These 26 measurements were divided into two data sets corresponding to two sweeps through the 13 one-third octave bands of interest. Data set one began with the 1 kilohertz one-third octave band and successively stepped up to the 16 kilohertz band. Conversely, data set two began with the 16 kilohertz band and successively stepped down to the 1 kilohertz band, thereby providing a check on the measurements of the data set one. Since the responses of eight hydrophones were recorded for each measurement, the number of decays recorded for each projector location was 208. Since the projector was moved to six different locations for each test condition, there were 1,248 decays recorded for each test condition. The total number of decays recorded for the four test conditions was, therefore, 4,992.

As verification of the linearity of the recording process, calibration signals were recorded on each reel of tape. These calibration signals consisted of the same type of signals used to excite the tank and were generated using the same equipment. Figure 5 is a block diagram of the calibration method. The signal, after being filtered into the desired one-third octave band, was fed through a decade attenuator to a junction box and then to the amplifier which had been used for the hydrophones. After adjusting the amplifier gain settings to match those used in the reverberation measurements for the frequency band being calibrated, the output of the noise generator was adjusted to match the signal level recorded during that measurement. The voltage of the signal was then measured using the B & K sound level meter and documented. This initial calibration signal was recorded on tape before the decade attenuator was used to decrease the signal level by 10 decibels. After measuring and recording this reduced level, the process was repeated two more times so that a total of four calibration levels were recorded with a 30 decibel difference between the first and the last level. This calibration test was recorded on each reel of magnetic tape for the frequency bands which had been used in the reverberation measurements recorded on that tape.

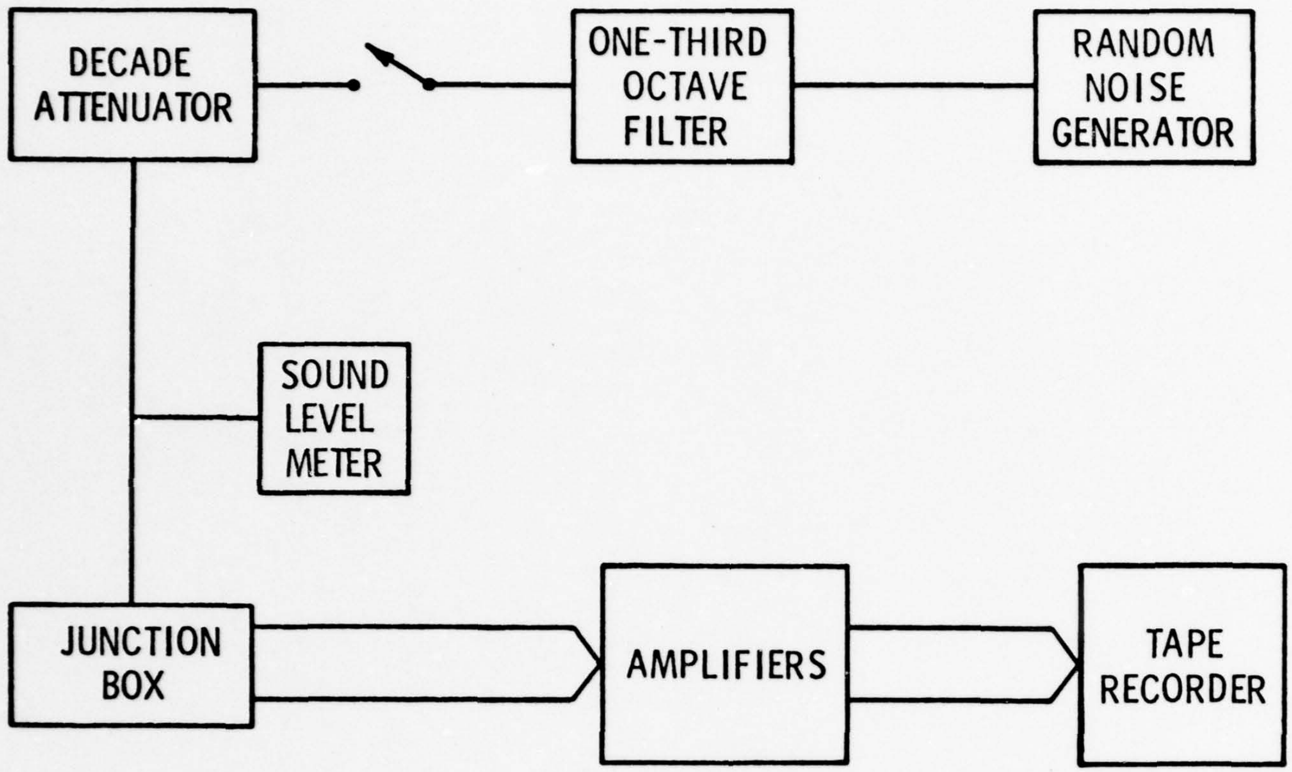

Figure 5. Block diagram of calibration method.

CHAPTER III

ANALYSIS METHODS

3.1 Preliminary Considerations

The general method of analysis for determining reverberation time from recordings of the decaying sound field in a room consists of filtering the recorded signal into the desired frequency band and recording the logarithm of the output on a graphic level recorder. From the record of the decay, the reverberation time is the time interval during which the recorded signal drops 60 decibels. If the decay does not extend over a range of 60 decibels, the reverberation time is determined by extrapolation using the slope of the linear portion of the decay.

In the procedure outlined in ANS S1.7-1970 Standard Method of Test for Sound Absorption of Acoustical Materials in Reverberation Rooms, the preferred method for limiting the test signal to one-third octave bands is for the pass-band of the sound source circuit to be wider than one-third octave and the pass-band of the receiver circuit to be exactly one-third octave wide. Figures 6 and 7 show typical spectra of the recorded data from test conditions one and three as measured using a Federal Scientific Ubiquitous Spectrum Analyzer (Model UA-500). In both figures, the upper curve represents the spectrum of the recorded signal without any filtering, while the lower curve represents the same signal after being filtered into the desired one-third octave band. Since the spectra of the unfiltered signals are

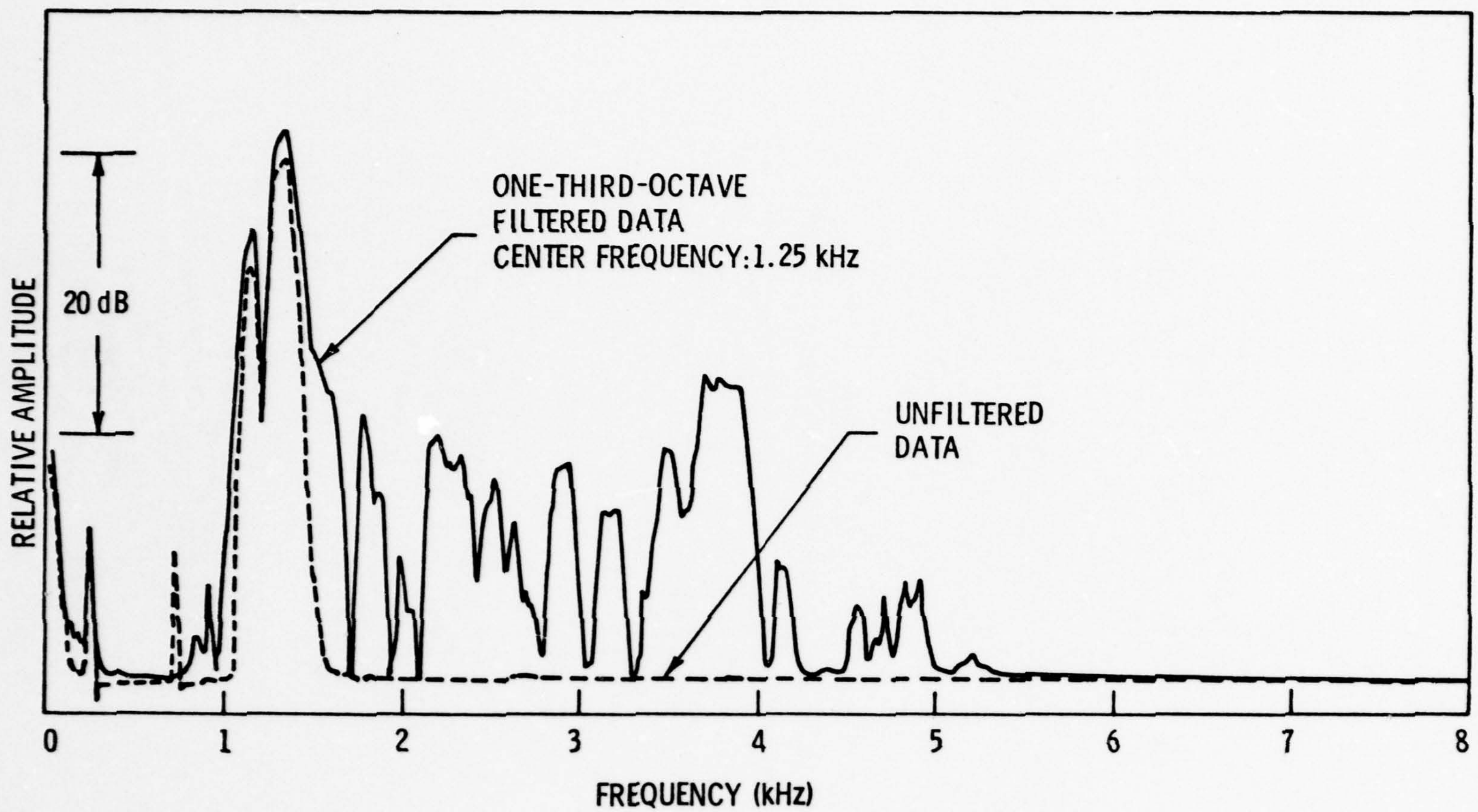

Figure 6. Typical spectra of recorded data from test condition one.

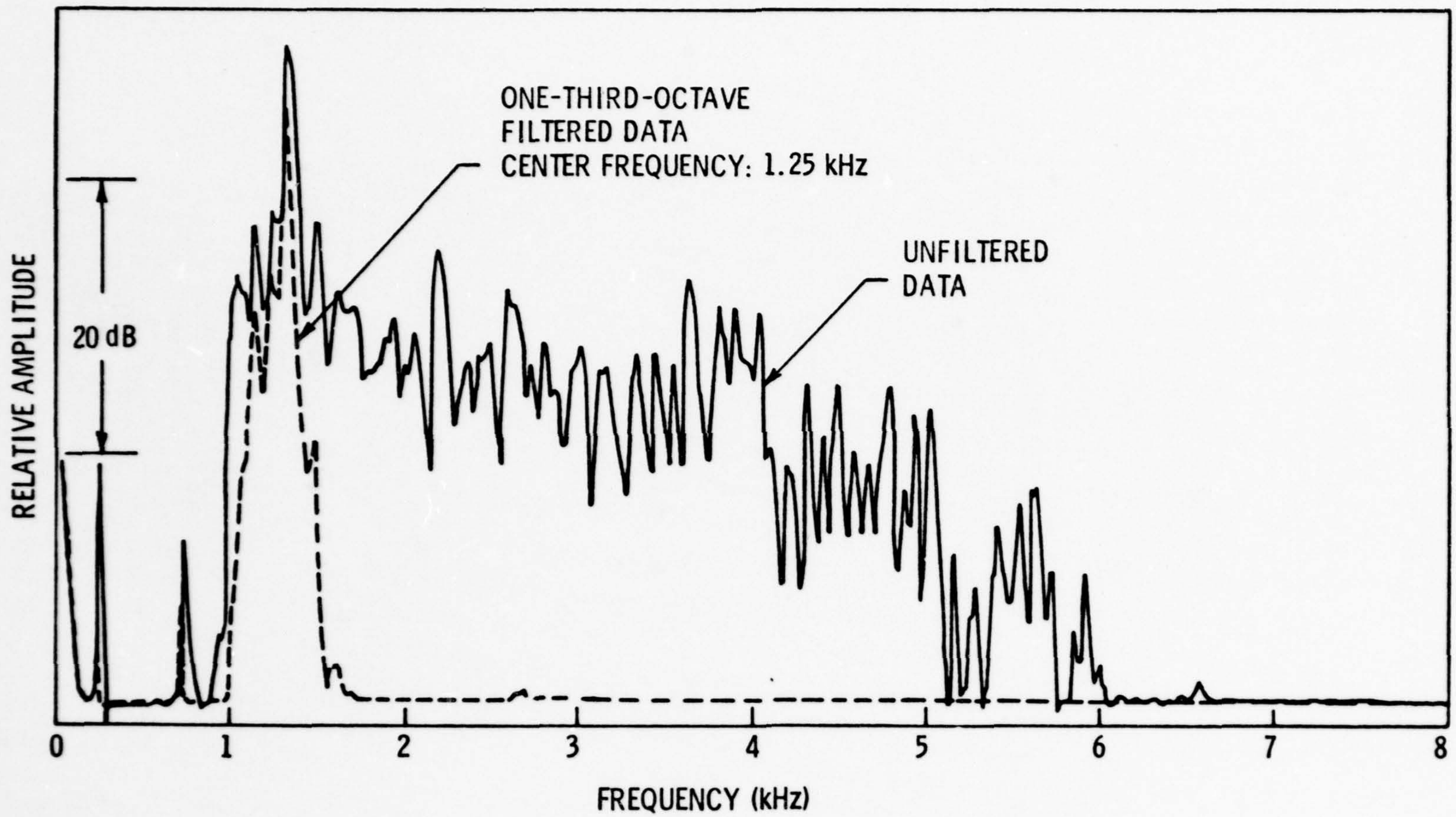

Figure 7. Typical spectra of recorded data from test condition three.

clearly much wider than one-third octave, any analysis method used for this data must include filtering the data into the appropriate one-third octave band.

Early in the preliminary analysis of the recorded data, it became evident that a major problem would be the response time of the equipment used in the analysis. As discussed previously, the differences between the acoustical properties of air and water cause the reverberation time of a chamber filled with water to be much shorter than when the same chamber is filled with air. As a result, equipment which normally responds fast enough for reverberation measurements in air was found to respond too slowly to be useful in underwater reverberation measurements. Several different analysis methods were, therefore, investigated in order to find a way to analyze the very rapid decay of the sound field in the MQL tank.

3.2 Analog Analysis Method

The first method of analysis investigated follows the previously outlined general method using analog equipment as indicated in the block diagram in Figure 8. Two different filter sets were used for this method, one by General Radio Company (Type 1925 Multifilter) and one by B & K (Audio Frequency Spectrometer Type 2112). The chart recorder was the Mark 240 Recorder of the Brush Instrument Division of Clevite Corporation (Model 2200-6607-10) equipped with Brush Logarithmic Amplifiers (Model 15-5221-11). The tape recorder used in this method as well as the other methods investigated was an Ampex Recorder (Model 1800L with ES-100 signal electronics).

Figure 8. Block diagram of analog analysis method.

The general procedure followed was to play back the recorded decay signal, filter it into the desired one-third octave band, and plot the decay on the chart recorder. Even though the chart recorder had a paper speed of 200 millimeters per second and a high pen speed, it was found that the decay was too fast to give usable plots on the chart recorder when the tape recorder was operated at the normal speed of 38 centimeters per second. In an attempt to alleviate this problem, the tape recorder was operated at speed reductions of 2 to 1, 4 to 1, and 8 to 1, while the center frequency of the filter was lowered by one, two, or three octaves, respectively. The equalization used in reproducing the recorded data was also adjusted to match the tape speed being used. The shapes of the decay curves produced by this technique showed significant improvement for determining the reverberation time from the slopes of the curves, particularly at the greater speed reductions. However, since the response time of the chart recorder in the analysis method was under investigation, it was decided that the response time of the rest of the equipment should also be investigated as a precaution.

To study the response time of the filter, the signal from a random noise generator was fed directly to the filter whose output was recorded on the chart recorder. The same toggle switch which had been used in the reverberation measurements was used to interrupt the signal between the noise generator and the filter. The ringing of the filter after the signal had been turned off was analyzed in the same manner as a decaying sound field so that a reverberation time was sought for each filter setting. At the higher filter settings, the filter responded

much faster than the chart recorder so that no reverberation time could be determined for those filter settings. At about 1.6 kilohertz, the filter began to show a response time longer than that of the chart recorder and, at about 800 hertz, fluctuations began to appear in the decay curves of the filter response as compared to the smooth, linear decays of the higher frequency settings. The response time became longer and the fluctuations greater as the frequency setting of the filter was lowered, thus raising the question of whether the decay curves plotted using the tape speed reduction technique were due to the decaying sound field in the tank or to the ringing of the filter being used. To carry the investigation further, a way was sought to eliminate the limited response of the chart recorder as a factor in the problem.

3.3 Hybrid Analysis Method

A way of effectively eliminating the problem of the limited response time of the chart recorder was found through the use of a transient store device by Physical Data, Inc. (Model 512A). This device can capture and store an analog signal by digitizing the input signal and storing it in a solid-state memory capable of holding 1,024 points of data. One of the methods of analog output from this device was a slow, single sweep for chart recorders or X-Y plotters. This method of display was used to overcome the limited response of the chart recorder by allowing the use of slower paper and pen speeds in the recording of the decay. Figure 9 shows a block diagram of how the transient store device was used. The method uses the same equipment as before with the transient store inserted between the filter and the chart recorder.

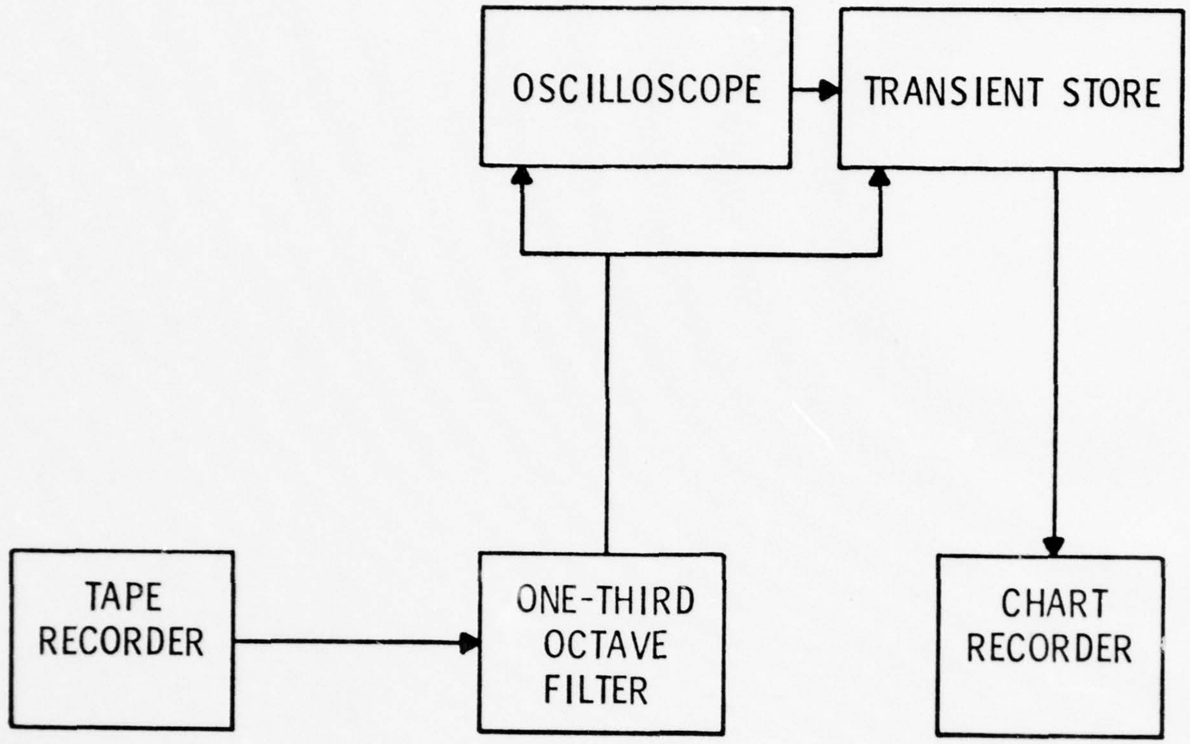

Figure 9. Block diagram of hybrid analysis method.

Triggering of the transient store can be accomplished by manual, internal, or external trigger. Because the manual trigger relies on human reaction time, it was unusable for capturing the short decays under study. The internal trigger, while suitable for the purpose, did not provide as much flexibility as an external trigger; therefore, an oscilloscope was used as an external trigger for detecting the decay. The transient store had a variable time base which permitted the storing of the analog signal in segments from 10 milliseconds to 100 seconds. In addition, a delay feature allowed the storage of the signal at set time intervals in relation to the trigger command. The settings for this feature ranged from -1 to +2 in intervals of one-half the time base, where -1 represents a pretrigger which is obtainable due to the fact that the trigger command stops the recording of data rather than initiating it. Under normal operating conditions, the analog signal being fed to the transient store is continually being digitized and recorded in memory with new data replacing the old until the trigger command stops the recording of data. It is, therefore, possible to store in memory a portion of the signal which preceded the trigger. This feature was extremely useful in the recording of the decay using the decay itself as the triggering event.

When this analog-digital system was used to investigate the recorded reverberation data, the plots showed fluctuations in both the excitation signal and the decay which were more rapid and pronounced than when the purely analog system was used. This was taken as an indication that the response time of the transient store was much faster than that of the chart recorder and, therefore, was more suitable for analyzing the short decays recorded in the MQL tank.

When the response time of the filter was investigated using this hybrid method, a frequency dependence was again found with the reverberation time of the filtered switch transient becoming longer for lower frequency settings of the filter. A comparison of the reverberation times of the filtered switch transient and the recorded data showed that at the lower frequencies, the reverberation times of the filtered switch transients were of the same order of magnitude as the reverberation times of the recorded data. Although the reverberation times of the filtered switch transients were generally shorter than those of the recorded data, the differences between the two were small enough to be within the range of variability of the data, particularly in view of the difficulties in obtaining consistency in determining the slope of a fluctuating decay curve using the human eye to perform the curve fit. Because of the conflict between the need to filter the recorded data into one-third octave bands and the slow response of the analog filters in the lower frequency range, another method of filtering the data was sought.

3.4 Digital Analysis Method

3.4.1 General Approach. The limitation on the transient response time of analog filters cannot be overcome using analog techniques. The transient response time of analog equipment is inversely proportional to the bandwidth of the equipment, and since the bandwidth of a one-third octave filter becomes narrower as the center frequency is lowered, the transient response time of the filter becomes longer with lower frequency. The only approach available to overcome this limitation is to use digital techniques.

Filtering using digital techniques can be accomplished by many different methods. A simple method which avoids the problem of limited transient response time is to approximate the desired filter characteristics using Fast Fourier Transforms. Filtering using FFT's is accomplished by performing a sequence of FFT's of the same size on the digitized data of interest. More explicitly, the procedure is to divide the digitized time information data into consecutive blocks of data corresponding to the size of the FFT being used. An FFT is performed on each block of data in the time domain to yield a block of data in the frequency domain. Each number in that block of frequency data represents the power in a specific segment of the spectrum of the digitized signal during the time interval of the block of data used by the FFT. Each segment of the spectrum is called a bin and is identified by a bin number. The frequency interval of the segment is called the bin width which is a constant for a given size FFT. Filtering of a signal can, therefore, be accomplished by considering the output of only those bins of the FFT which match the desired filter characteristic. By considering only the output of one bin when taking a sequence of FFT's, a signal can be filtered into a frequency band specified by the bin width, the bin number, and the FFT size.

Because the Fourier Transform is a process which trades time information for frequency information, the resolution of the information in the time domain is inversely related to the resolution of information in the frequency domain. Thus, achieving good resolution in the frequency domain by requiring a narrow bin width dictates a large size FFT which results in poor time resolution. Similarly, good

time resolution is possible using a small size FFT having a wide bin width which results in poor frequency resolution. Since reverberation time measurements require a knowledge of both time and frequency information, a judicious selection of FFT size is needed to yield a satisfactory approximation to one-third octave filtering while retaining adequate time resolution. The FFT's selected for use in the digital analysis of the MQL reverberation measurements are listed in Table I. The listing includes the widths and limits of the FFT bins and the one-third octave bands for comparing the approximations made to achieve digital filtering.

It should be noted that for the larger FFT's, the time resolution obtainable using the approach outlined above is only marginally adequate for reverberation measurements. To increase the time resolution for the larger FFT's and still retain the same filtering effect, the blocks of data can be divided in half so that an FFT can be taken on each pair of consecutive half-blocks of data. To maintain some degree of independence between consecutive FFT's, the input data is weighted using a set of coefficients called a Hamming window or raised cosine function given by the equation:

$$C_J = 0.54 - 0.46 \cos 2\pi \left(\frac{J - 1}{N - 1}\right) \quad ,$$

where N is the size of the FFT, and J is a number varying from one to N corresponding to the index number of the data point which will be weighted by that coefficient. Figure 10 shows the shape of the Hamming window used to weight the unfiltered data. The use of this technique employing the Hamming window allows twice as many FFT's to be taken on a sequence of data, thereby increasing the time resolution by a factor of two.

TABLE I

ONE-THIRD OCTAVE BAND APPROXIMATIONS USING FAST FOURIER TRANSFORMS

Center Freq. (kHz)	Band Width (Hz)	Band Limits (Hz)	Approx. BW (Hz)	FFT Size	FFT Type	Bin No.	Bin Limits (Hz)	Time Res. (ms)	Pts. 200ms	Hamming Window	Pts. 200ms	Time Res. (ms)
1	231	891-1122	200	256	Even	6	900-1100	5	40	Yes	80	2.5
1.25	291	1122-1413	400	128	Even	4	1000-1400	2.5	80	Yes	160	1.25
1.6	365	1413-1778	400	128	Even	5	1400-1800	2.5	80	Yes	160	1.25
2	461	1778-2239	400	128	Even	6	1800-2200	2.5	80	Yes	160	1.25
2.5	579	2239-2818	800	64	Even	4	2000-2800	1.25	160	No		
3.15	730	2818-3548	800	64	Even	5	2800-3600	1.25	160	No		
4	919	3548-4467	800	64	Even	6	3600-4400	1.25	160	No		
5	1156	4467-5623	1200	128	Odd	12,13,14	4400-5600	2.5	80	Yes	160	1.25
6.3	1456	5623-7079	1600	32	Even	5	5600-7200	0.625	320	No		
8	1834	7079-8913	1600	32	Even	6	7200-8800	0.625	320	No		
10	2307	8913-11220	2400	64	Odd	12,13,14	8800-11200	1.25	160	No		
12.5	2910	11220-14130	3200	32	Odd	8,9	11200-14400	0.625	320	No		
16	3650	14130-17780	3200	32	Odd	10,11	14400-17600	0.625	320	No		

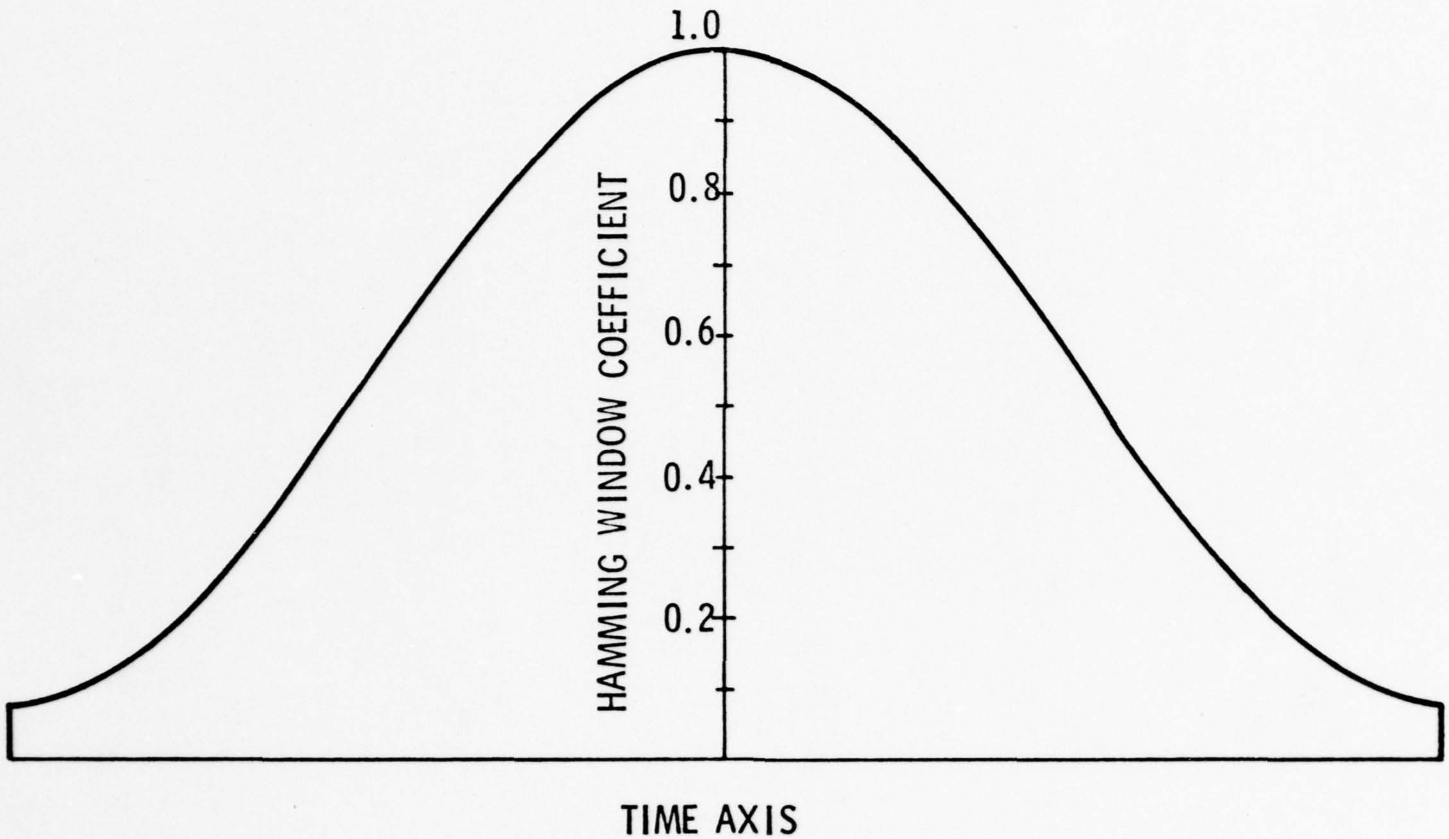

Figure 10. Plot of Hamming window or raised cosine function.

By using digital techniques to overcome the limited transient response time of analog filters, an additional benefit results which provides a consistent method for determining the slope of the decay in reverberation time measurements. When analog techniques are used, the slope of the decay is determined by manually drawing a straight line through the linear portion of the decay curve and measuring the slope of that line using a protractor or some other geometric technique. The selection of the straight line which best fits the decay is subject to considerable human variability. When large numbers of decay records have to be processed, human fatigue can cause serious inconsistencies in the selection of the best fit from one decay to the next. In addition, fatigue can affect the accuracy of the method for determining the slope of the line which best fits the decay after it has been chosen. At its best, the manual method is adequate, but its lack of consistency in processing large quantities of data is a severe disadvantage.

With the reverberation data in digital form, however, the problem of consistency in the determination of the slope of the decay can essentially be eliminated through the use of a linear, least-squares curve fit. This method has no difficulty in handling the fluctuations in the decay which are a possible source of error in the manual method. The consistency of this method is clearly demonstrated by the fact that a unique slope is calculated for a given set of data. Because of the availability of the linear, least-squares curve fitting method, the problem encountered in the digital determination of the slope of the decay is not the selection of the best fit to the data as

in the manual method but rather the selection of the data over which the curve fit will be performed. This problem, by its very nature, is much easier to solve than that of manually selecting the best fit since the decay is bounded by the excitation signal and the background noise. An algorithm can, therefore, be written to locate the decay by determining which data points lie between the excitation signal and the background noise. Essentially, the algorithm located the top and bottom of the decay so that the linear, least-squares curve fit can be performed on the set of data enclosed by these two points. When properly implemented on a digital computer, the algorithm can be consistently applied to large numbers of data records without human intervention.

3.4.2 Digital Data Preparation. Since the reverberation measurements in the MQL tank were recorded using an analog tape recorder, the analog data had to be converted into digital form before the digital analysis techniques could be used. Figure 11 is a block diagram of the method used in the analog to digital data conversion. The analog tape recorder used to reproduce the analog data tapes was the same recorder which had been used in the previously described analog analysis method. The low-pass, anti-aliasing filter was a variable electronic filter by Spencer-Kennedy Laboratories, Inc. (Model 302). The tape recorder used as an analog delay line was made by Ampex Corporation (Model FR 1300 with ES-100 signal electronics). The A to D converter was a Datel Systems, Inc. 10-bit converter (DAS-16 Series) which was controlled using circuitry designed and built at the Applied Research Laboratory. The frequency synthesizer used to control

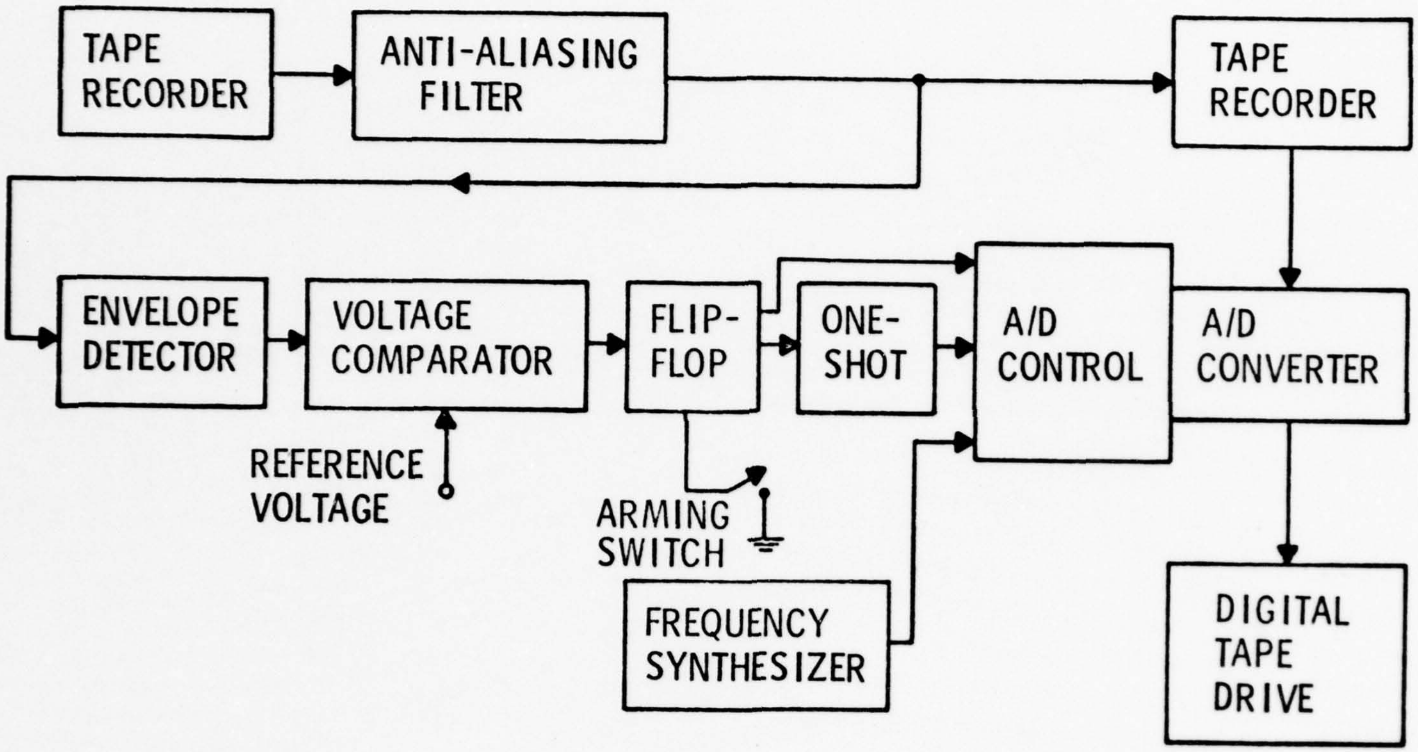

Figure 11. Block diagram of analog to digital data conversion.

the sampling rate was made by General Radio Corporation (Type 1163-A). The digital tape drive was made by Pertec (Model T9660-6-75). The digitized data was recorded on 1.27 centimeter wide IBM Multi-System Magnetic Tape.

One of the most important decisions that had to be made before any analog data could be digitized was the choice of a sampling rate for the digitizing process. To adequately sample an analog signal, the sample rate must be at least twice as fast as the highest frequency signal of interest. The upper band limit of the 16 kilohertz one-third octave band is 17,780 hertz, but the spectra of the recorded data measured during the preliminary analysis showed recorded analog signals out to about 25 kilohertz. The sample rate necessary for a 25 kilohertz signal is on the order of 50,000 samples per second. For convenience in digital computations, it is often useful to choose a sample rate which is proportional to some power of two. In this case, a sample rate of 51,200 samples per second was chosen so that the bin widths of the FFT's would be in round numbers. To achieve such a high sampling rate, the analog tape was played at a speed reduction of 4 to 1, while the digitizer was operated at a sample rate of 12,800 samples per second. To prevent possible errors in sampling, the analog signal was fed through a low-pass, anti-aliasing filter before being sent to the digitizer. This low-pass, anti-aliasing filter prevented signals higher than one-half the sample rate from being misinterpreted by the digitizing process as lower frequencies which would interfere with the data of interest. The upper limit of the anti-aliasing filter was set at about 6 kilohertz so that in real time, the effective cut-off

frequency would be about 24 kilohertz. Two sections of the Spencer-Kennedy variable electronic filter were used to give an attenuation of about 36 decibels per octave above the cut-off frequency.

To facilitate locating the decay, the analog tape was played in reverse at 9.5 centimeters per second, and only one channel of data was digitized at a time. After having been fed through the anti-aliasing filter, the analog signal was recorded on a second tape recorder which served as an analog delay line. The input to this recorder was monitored with circuitry for detecting the decay. This circuitry consisted of an envelope detector, a voltage comparator, a flip-flop (bi-stable multivibrator), and a one-shot (mono-stable multivibrator). The output of the envelope detector yielded a DC voltage proportional to the waveform envelope of the AC analog signal. By using the voltage comparator to compare the output of the envelope detector against a reference voltage which had been chosen to be just above the envelope of the background noise of the analog signal, the recorded decay could be located as it was being played in reverse.

When the decay had been detected, a signal from the voltage comparator caused a flip-flop to change state and trigger a one-shot which served to control the length of time the digitizer operated. The flip-flop was equipped with a momentary arming switch to reset the flip-flop in preparation for the next decay to be digitized. This provided a manual control to prevent accidentally triggering the digitizer when the tape recorder was started or stopped and to prevent the retriggering of the digitizer by fluctuations in the excitation signal being played back after the decay had been digitized. The one-shot controlled the digitizer through TTL circuitry. This

circuitry kept the digitizer on as long as a voltage interpreted as a one by the logic circuits appeared at the input to the digitizer controller. If a voltage appearing at the controller input was interpreted as a zero, the digitizer was turned off. Because the output of the one-shot was TTL compatible, the one-shot's stable state turned off the digitizer, while its unstable state turned the digitizer on. By adjusting a potentiometer on the one-shot, the length of time the one-shot remained in its unstable state could be varied, thus controlling the length of time the digitizing process took place. Since the analog tape was reproduced at a speed reduction of 4 to 1, the length of time the one-shot remained in its unstable state was set at about 1.6 seconds so that in real time, about 400 milliseconds of data were digitized for each decay.

While the above control process was taking place, the analog signal was being recorded on another tape recorder which served as an analog delay line. Because the record and playback heads were separated by 8.89 centimeters, this recorder was operated at 9.5 centimeters per seond to give a delay of about 933 milliseconds to the analog signal. This delay enabled the controlling circuitry to detect the decay and turn on the digitizer before the decay signal appeared at the input to the A to D converter. Because of the speed reduction and the reversal in direction of the playback of the tape, the effect of the delay was to permit the digitizing of about 400 milliseconds of data containing approximately 166 milliseconds of the excitation signal and decay, and 233 milliseconds of the background noise after the decay. After the decay was digitized, it was recorded

on magnetic tape using a 9-track, phase-encoded digital tape drive. Each sample of digitized data was encoded with an identification number. This identification number was automatically incremented each time the decay detection circuitry was armed by resetting the flip-flop. In addition, the digitized data for each decay was delineated as a file on the tape by a coded signal called a file mark. By noting the number of files recorded before a particular decay along with the identification number recorded with the data, each digitized decay was uniquely identified for later retrieval.

3.4.3 Computer Data Analysis. The analysis of the digitized MQL reverberation data was performed using a Fortran IV computer program. Figure 12 is a block diagram of the hardware used in the analysis of the data. The 16-bit CPU (Model 7/16), 64k-byte memory core, and the two cassette tape drives were made by Interdata, Inc. The Pertec digital tape drive was the same one used in the digitizing of the analog data. The interface and formatter for the Pertec tape drive were designed and built at the Applied Research Laboratory. The graphic computer terminal was made by Tektronix, Inc. (Model 4002A) along with the hard copy unit (Model 4601).

The basic operations performed by the analysis program, Reverb, were locating the decay in the file of digitizing data, filtering the digitized data using FFT subroutines, identifying the top and bottom of the decay, and calculating the reverberation time using the slope of the decay as determined by a linear, least-squares curve fit. Because of the size of the program and the limited amount of core memory available, the program had to be divided into a root segment which

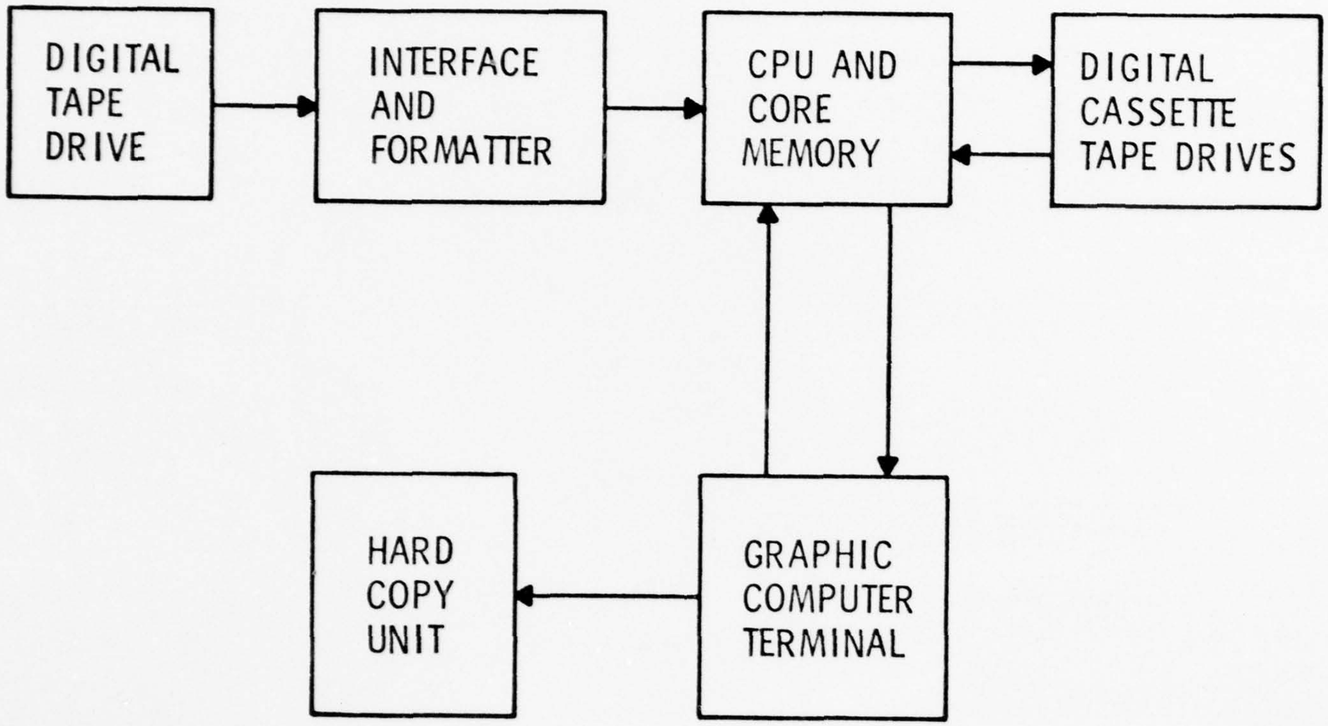

Figure 12. Block diagram of computer system hardware.

remained in core and three overlays which were stored on a magnetic tape cassette and brought into core by the root segment as needed. Each overlay consisted of a primary subroutine, identified by the order of the overlay in the sequence of the analysis, plus any other subroutines needed to perform the calculations in the primary subroutine. These secondary subroutines either were specially written for this specific analysis program or were taken from the general systems software library supplied with the computer. Figure 13 is a simplified flow chart of the root segment and overlay one. Figures 14 and 16 are flow charts of overlay two and overlay three. Figure 15 is a flow chart of subroutine Redat which was specially written for use in the second overlay of this program. (A complete listing of program Reverb has been documented and made available for reference.)

Program Reverb was designed for either manual or automatic operation. In the program's manual mode, the operator exerted his control over the analysis process by typing replies on the computer terminal's keyboard to questions written on the terminal's screen by the program asking for information needed for the analysis or for a decision by the operator as to what course the program should take. This interactive approach allowed great flexibility in the operation of the program since the operator could choose to reserve for himself either the maximum number of decisions required during the analysis or the minimum number of decisions or any number in between with the remainder being performed according to fixed algorithms implemented in the program.

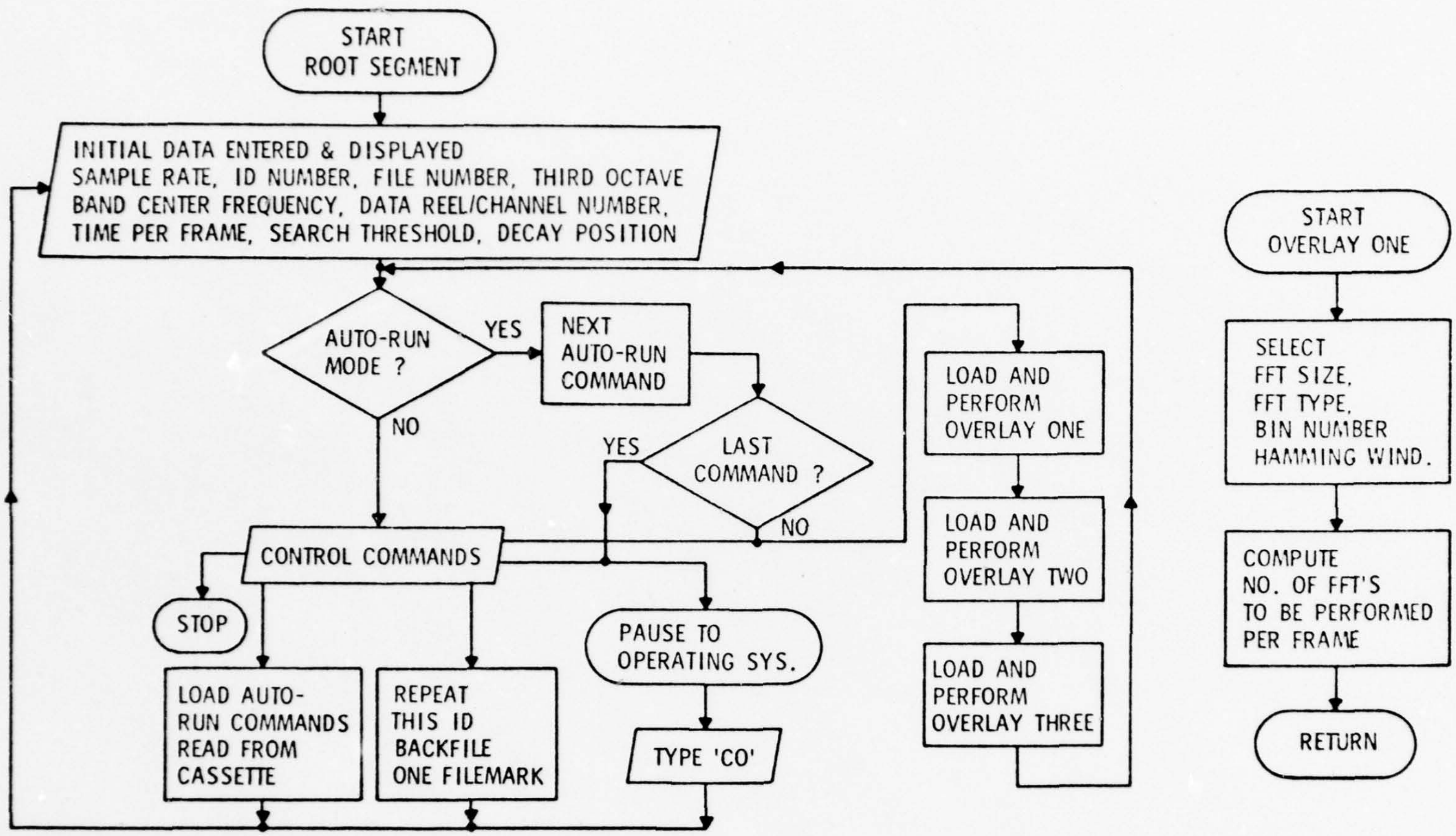

Figure 13. Simplified flow charts of root segment and overlay one of analysis program Reverb.

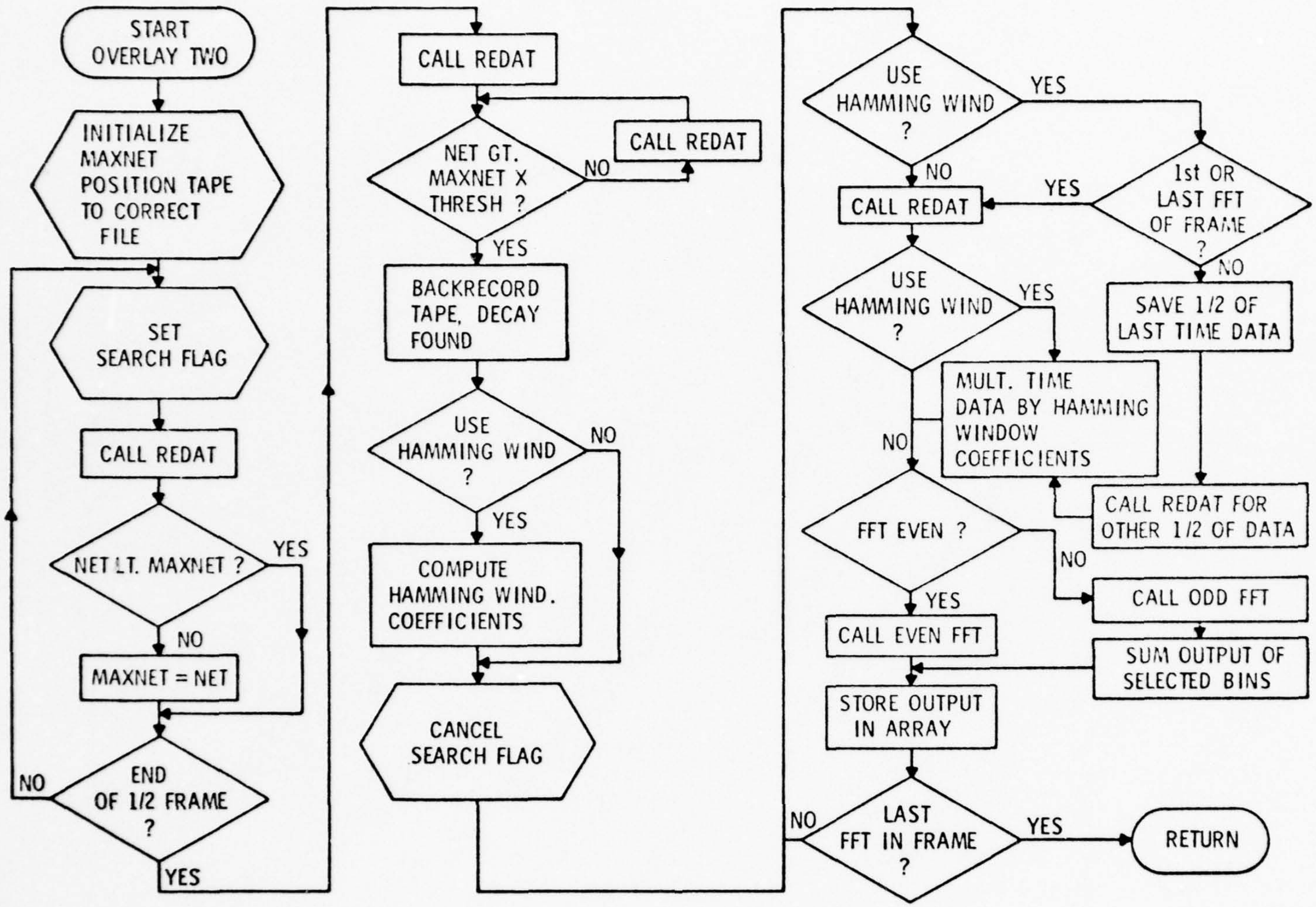

Figure 14. Simplified flow chart of overlay two.

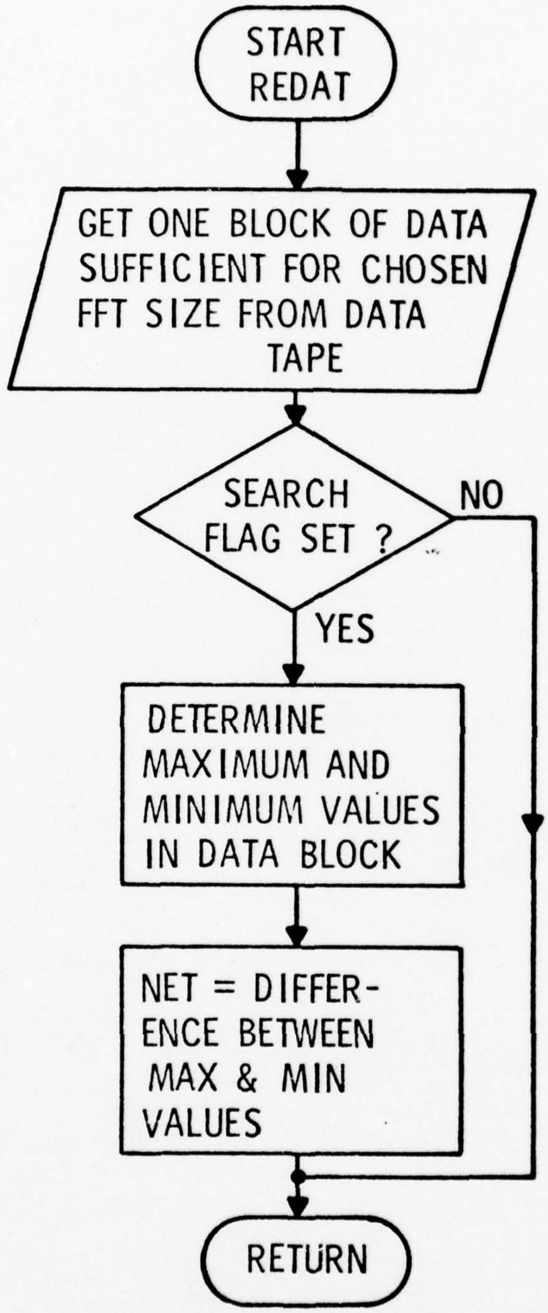

Figure 15. Simplified flow chart of subroutine Redat.

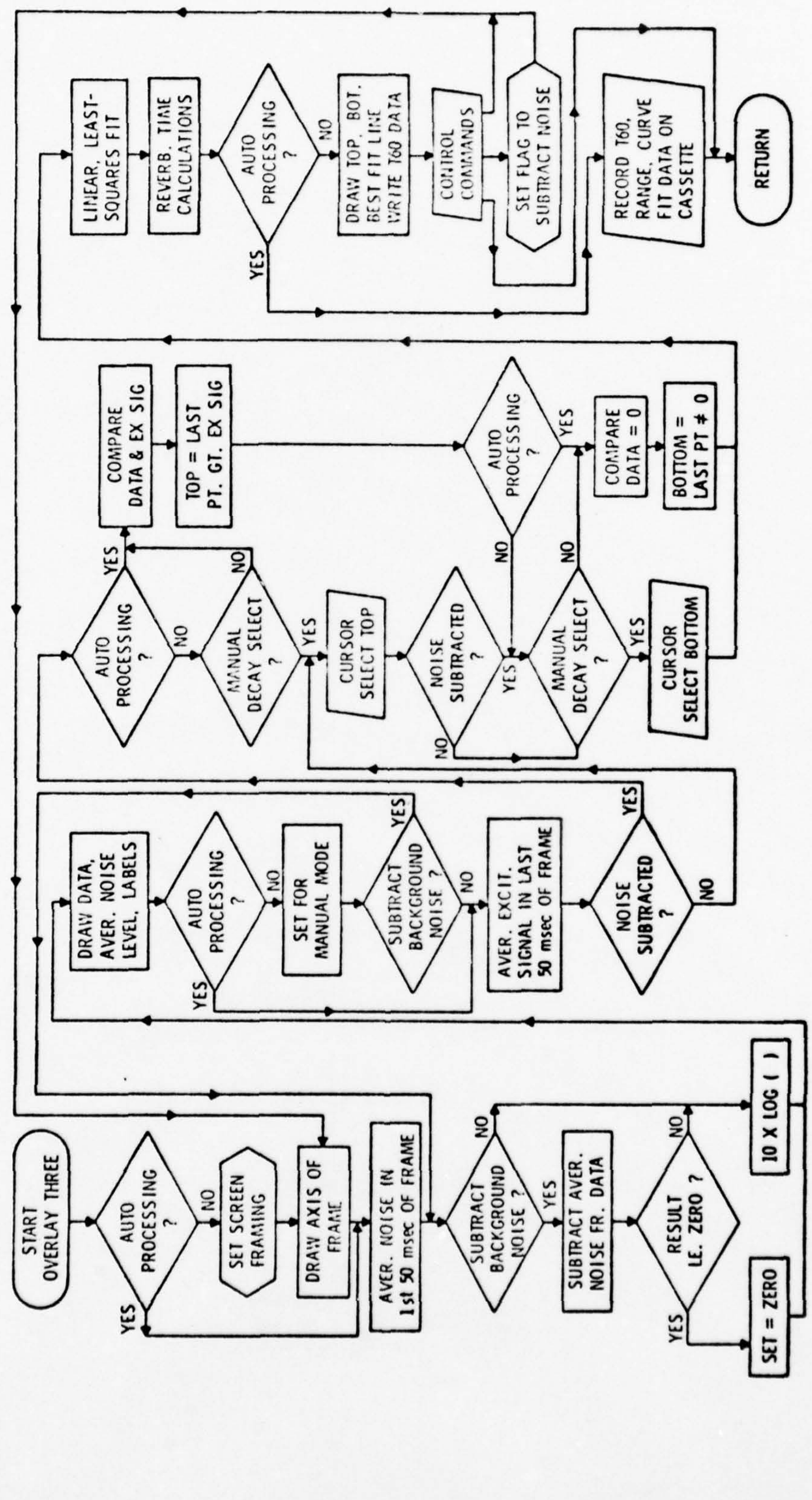

Figure 16. Simplified flow chart of overlay three.

In its automatic operation, the program required a list of commands to be stored in memory before the automatic decay analysis could begin. These commands essentially consisted of the information needed by the program for making decisions without operator assistance. The algorithms used in this automatic run mode were the same ones used in the manual mode. The major difference between the auto-run and manual modes was that all the opportunities for operator input in the manual mode were bypassed in the auto-run mode. An additional difference between the two modes of operation was the manner in which the input and output of information took place. In the manual mode, the program asked questions and output data on the screen of the graphic terminal while the operator controlled the analysis by typing commands and information on the keyboard of the terminal. In the auto-run mode, the required list of commands was read by the program from a cassette which had previously been prepared by the operator using the system software supplied with the computer. Once the list of commands had been read and stored in memory, the program began the automatic analysis of the digitized reverberation data. Upon completion of each analysis, the program recorded the results on a blank cassette for later retrieval. When the final command in the list had been performed, the program recorded a file mark on the blank cassette indicating the completion of the auto-run. The operation of the program was then automatically terminated. In this manner, no input or output of data took place in the auto-run mode using the screen or keyboard of the graphic terminal.

As mentioned previously, program Reverb was divided into a root segment and three overlays. The purpose of the root segment was to

permit the input of information needed for the data analysis and to control the overall operation of the program. All analysis calculations were performed by the overlays using information derived from parameters defined in the root segment. Certain parameters such as the sample rate were never changed during this analysis, but the option for changing them was included to make the program more versatile for general use. The ID number and file number were parameters for locating the digitized data on the magnetic tape and thus were changed for each decay that was analyzed. Similarly, the third-octave band center frequency was a parameter which was changed quite often since it controlled how the data was filtered using the FFT subroutines. Other parameters either controlled the acquisition and graphic presentation of the decay on the screen of the terminal or provided information for identifying the results of the digital analysis with the analog recordings of the reverberation measurements.

The search threshold parameter controlled the process of acquiring the decay from the digitized data in the file. This parameter was used to determine the reference value against which the digitized data was compared in the algorithm for locating the decay. When the algorithm found data which equaled or exceeded this reference value, it was decided that the decay had been found.

The time-per-frame parameter controlled the interval of time represented on the screen of the terminal whose length was defined to be one frame. Control of this parameter is of particular importance when dealing with decays recorded under different test conditions. A long decay requires a longer time per frame than a short decay so that it can be plotted in its entirety. While it is not necessary to the

analysis that the time per frame used for a short decay be shortened from that used for a long decay, it is economically wise to do so since the time per frame is one of the parameters used to determine how many FFT's must be performed which largely determines the execution time of the program. In the preliminary testing of the program, it was found that test condition one required a time per frame of 200 milliseconds, while test condition three required 240 milliseconds.

The decay-positioning parameter is also important for the graphic presentation of the results. This parameter was used to center the decay in the frame so that the plot would show the entire decay. In general, this parameter was only changed if the time per frame was changed since the change in the time scale also changed the time relation for the center of the frame.

The final parameter was the data reel/channel number which was a coded number identifying the analog tape reel and channel from which the data had been digitized. This information served to document the link between the results of the digital analysis and the analog measurements which had been made in the MQL tank. Beyond the identification purposes, the data reel/channel number had no significance in the analysis calculations.

As mentioned previously, all the analysis calculations were performed in the overlays which functioned as subroutines in relation to the root segment. The overlays in turn used subroutines in the analysis calculations, the most important of which were the FFT routines. The FFT routines used in this analysis were not part of the system software supplied with the computer, but rather were part of the general support routines which had been written by users of the computer at the

the Applied Research Laboratory. The FFT subroutines used to generate the power spectrum of the digitized data were of two types, even and odd. The essential difference between the even and odd FFT routines was the shift in frequency of the odd FFT bins by one-half the binwidth with respect to the even FFT bins. This shift in the FFT bins allowed a much better approximation of the one-third octave bands by the FFT method than would have been otherwise possible.

The selection of the parameters for the one-third octave band approximation was made in overlay one using the number specified in the initial data as the third-octave band center frequency. Through the selection process of overlay one, the FFT size, type, and bin number were automatically chosen along with the option to use the Hamming window on the unfiltered data for increasing the time resolution of the resultant decay plot. Overlay one also had a provision for manual override of the automatic selection process so that the program could be used to filter data into bandwidths other than one-third octave wide. This override capability makes the program more versatile for general use and testing purposes. Once the parameters for the filtering process had been chosen, the number of FFT's that must be performed per frame was calculated, thereby completing the operations of overlay one.

The portion of the analysis calculations concerned with finding the decay and filtering the data took place in overlay two. To begin the search for the decay, the second overlay positioned the tape with the digitized data to the correct file as specified in the initial data. The procedure for finding the decay in the file was

based on the fact that the data containing the decay was entered in reverse time sequence so that the background noise was read before the excitation signal. By determining the maximum swing in the data which occurred in the first half-frame of the file, a measure of the background noise was obtained which was multiplied by the search-threshold parameter to give a reference value against which the rest of the data could be compared. The algorithm for locating the decay proceeded by comparing the swing in consecutive blocks of data until a block of data was found where the swing equaled or exceeded the reference value. When this occurred, the decay was declared to have been found. The acquisition of the data from the file and the determination of the swing in each block of data were performed in the subroutine Redat while the comparison of the swing in the data with the reference value was performed in overlay two.

Subroutine Redat was used in several places in overlay two. Its main functions were to read the data from the tape, check that the ID number found corresponded to that specified in the initial data, and to check for errors in the digitized data. In addition, when the search for the decay was being conducted, the maximum and minimum values of the data within each data block were found, and a net value was obtained by subtracting the minimum value from the maximum value. This net value, representing the maximum swing in a block of data, was the measure of the data used for locating the decay as described in the previous paragraph.

When the decay had been found by reading consecutive blocks of data in the file until the net value exceeded the reference value, the data tape was backed up the number of data records specified by the

decay-positioning parameter so that the filtering process would be performed on the entire decay. Once the repositioning of the tape was completed, the filtering of the data began using the FFT size, type, and bin number specified in overlay one. The single number output from each FFT, representing the filtered time data in a particular interval of time, was stored in an ordered array after each iteration of the process until the necessary number of FFT's had been performed for the chosen time per frame. The completion of the analysis calculations in overlay two coincided with the completion of the last FFT in the sequence.

The final process in the analysis of the reverberation data was performed in overlay three whose principal function was the determination of the reverberation time from the slope of the filtered decay. As the most interactive portion of the analysis program, overlay three has a very complicated system of logic as can be seen in even the simplified flow chart in Figure 16. In its manual mode, overlay three presented a plot of the decay on the terminal screen to assist the operator in making decisions. Its simplest operation, however, occurred when the program was in its auto-run mode.

In auto-run, all drawing operations on the graphic terminal were suppressed. The algorithm for determining the top and bottom of the decay began by determining the average background noise in the first 50 milliseconds of the frame. This average noise value was then subtracted from all the data in the frame. This was done to lessen the effects of the background noise on the slope of the lower portion of the decay where the levels of the decaying sound field and the

background noise are nearly the same.[2] If the difference between a data point and the average noise value was zero or less, the resultant value was set at zero; otherwise, ten time the logarithm of the difference was taken. In this manner, data values less than the average noise value were ignored, data values very nearly equal to but still greater than the average noise value were greatly reduced, and data values much larger than the average noise value were essentially unchanged. To find the top of the decay, the average value of the excitation signal in the last 50 milliseconds of the frame was determined and used as a reference value. A comparison was then made between the reference value and each data point beginning at the point in time 50 milliseconds from the end of the frame and progressing to the beginning of the frame. The last data point found by that process to be greater than the average excitation signal was judged to be the top of the decay.

The search for the bottom of the decay began at the top of the decay and again progressed in a reverse direction toward the beginning of the frame until a data point with a value of zero was encountered. Since a value of zero indicated that at that point the level of the decay had dropped to or below the average background noise, the data point which preceded that zero value in the search sequence was judged to be the bottom of the decay. With the top and bottom of the decay thus identified, the reverberation time was calculated in a straightforward manner from the slope of the decay as determined by a linear, least-squares curve fit performed on the data between the top and the bottom of the decay. In the final operation of overlay three, the

results of the automatic data analysis were recorded on a blank cassette for later retrieval.

After the analysis program had been run in its auto-run mode, the results of the analysis which had been stored on a cassette were displayed using a small program called Retrev. (A complete listing of Retrev has been documented and made available for reference.) The sole function of Retrev was to read the results from the cassette and display them on the terminal screen so that a hard copy could be made. If an error had occurred in the analysis of a particular decay during the auto-run, the program Reverb would write a coded error message on the cassette in place of the results from that decay so that when Retrev was used, the error would be detected and easily interpreted. Except in extreme cases when the digitized data was totally unusable due to improper sequencing during the digitizing process, the problem which caused the error could be overcome using program Reverb's manual mode. The only solution for the extreme case was to redigitize the data from the analog tape.

Beyond the errors in the data due to improper digitizing, more subtle problems which had occurred in the auto-run mode could be detected by comparing the results obtained from data set one with those from data set two along with the results obtained at other hydrophone locations for the same frequency band. Results which deviated from the prevailing trend were investigated using the manual mode of operation. Due to the random nature of the excitation signal, the actual top of the decay may sometimes occur below the average level of the excitation signal. In a case such as this, the

algorithm's choice for the top of the decay yields a value for the reverberation time much higher than the true value. Similarly, fluctuations in the decaying sound field may sometimes be large enough to drop below the average background noise before the bottom of the decay has been reached. If this occurs well before the actual bottom of the decay, the value of the reverberation time will be much lower than the true value. Both of the problems just mentioned are easily detected by studying the results from an auto-run. When using the manual mode to investigate these problems, the operator can very quickly determine the cause of the problem and correct for it by using the interactive capabilities of the program.

When the program was operated in the manual mode, overlay three presented the operator with many options from which to choose. One option was to let the program follow the algorithm described above for finding the decay and determining the reverberation time but with the drawing operations no longer suppressed. Alternatively, the operator had the option of choosing either the top or the bottom of the decay and letting the program follow the algorithm in choosing the one that remained. To exercise this option, the operator had to activate a cursor on the graphic terminal screen controlled by a joystick attachment to the terminal. If he had opted to choose the top of the decay, the operator had to position the cursor to the point which he desired and then type the letter "A" on the terminal keyboard to indicate that the choice had been made. Similarly, for the bottom of the decay, he again had to position the cursor as desired, but the letter "B" had to be typed to indicate that the bottom on the decay had been chosen. Upon completion of either of these options, the

operation of the overlay continued as before except that the results of the curve fit and reverberation time calculations were displayed on the graphic terminal screen as a line with the calculated slope drawn next to the decay curve with the numerical results also written on the screen.

A final option which was also available to the operator was the complete manual selection of the top and bottom of the decay. When this option was chosen, the average background noise was not subtracted and the algorithm for locating the top and bottom of the decay was not functional. The linear, least-squares curve fit and the reverberation time calculations, however, were used. The selection of the top and bottom of the decay was performed using the cursor controlled by the joystick as previously described. Once the top and bottom of the decay had been manually chosen, the linear, least-squares curve fit was used to determine the slope of the decay from which the reverberation time was calculated. The results were presented on the screen as a line with the calculated slope drawn next to the decay curve with the numerical results also written on the screen. After the results had been displayed, the operator had the option of repeating any of the manual decay processing operations or returning to the root segment of the program.

CHAPTER IV

RESULTS

4.1 Graphic Results

Typical results from the analysis program Reverb are shown in Figures 17, 18, 19, and 20. Figures 17, 18, and 19 are typical results from test condition one, while Figure 20 is typical of test condition three. Figure 17 is a plot of the filtered data before any decay processing has been performed. Figure 18 is a plot of the same filtered data after the background noise has been subtracted, the algorithm for locating the top and bottom of the decay has been performed, and the reverberation time has been calculated using the slope of the decay as determined by the linear, least-squares curve fit. The top and bottom of the decay are represented in the plot by short, horizontal lines emanating from the points chosen by the algorithm. The dashed line next to the decay has the slope which was calculated using the linear, least-squares curve fit. It has been offset from its true position so as not to obscure the plot of the decay. The average values of the background noise and excitation signal are represented in the plot by horizontal lines emanating from the beginning and end of the frame, respectively. The length of each of these lines indicates the 50 millisecond interval over which the averages were taken.

Figure 19 is a plot of the same decay after it has been analyzed using the manual mode of program Reverb. As can be seen in

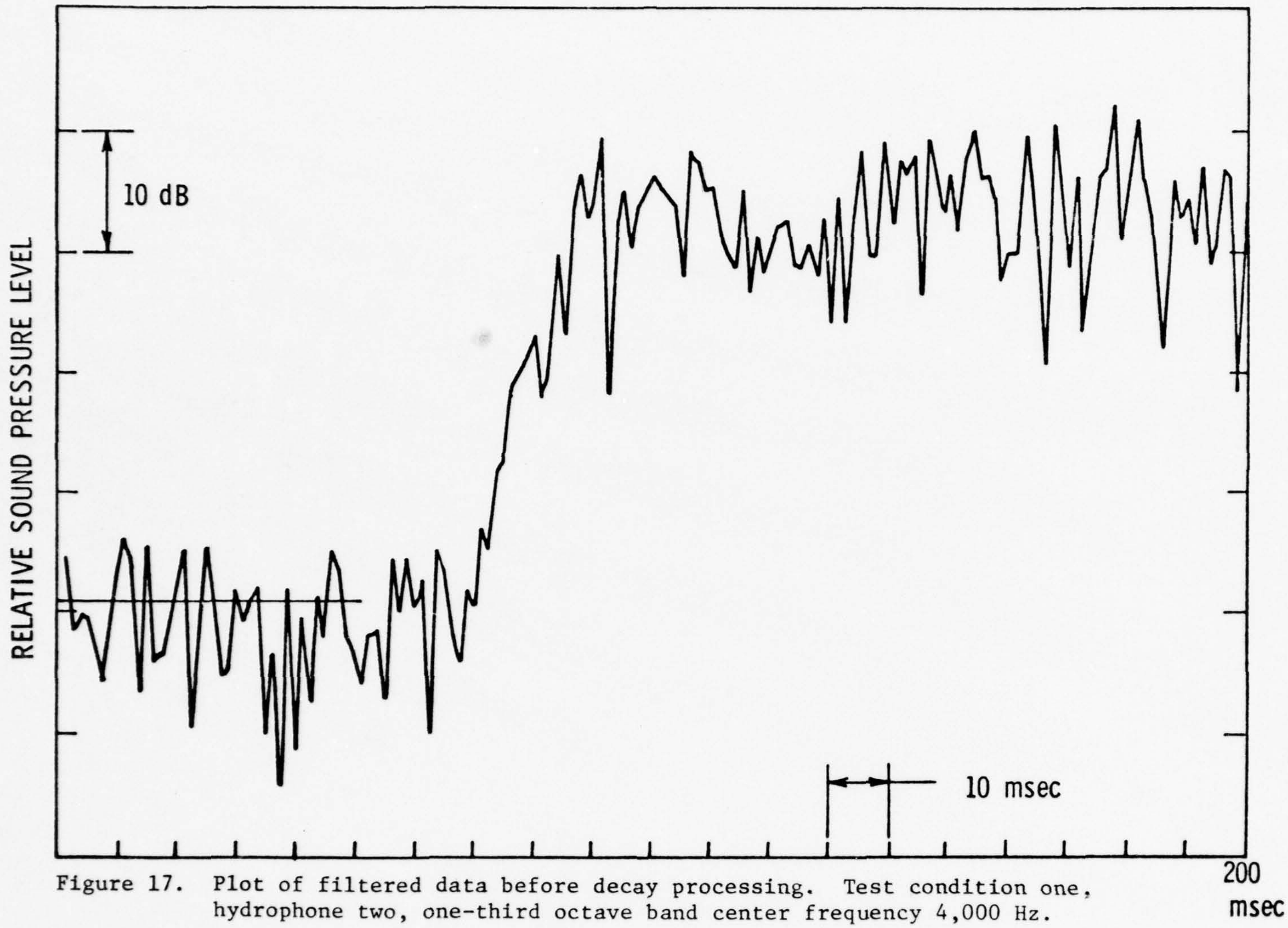

Figure 17. Plot of filtered data before decay processing. Test condition one, hydrophone two, one-third octave band center frequency 4,000 Hz.

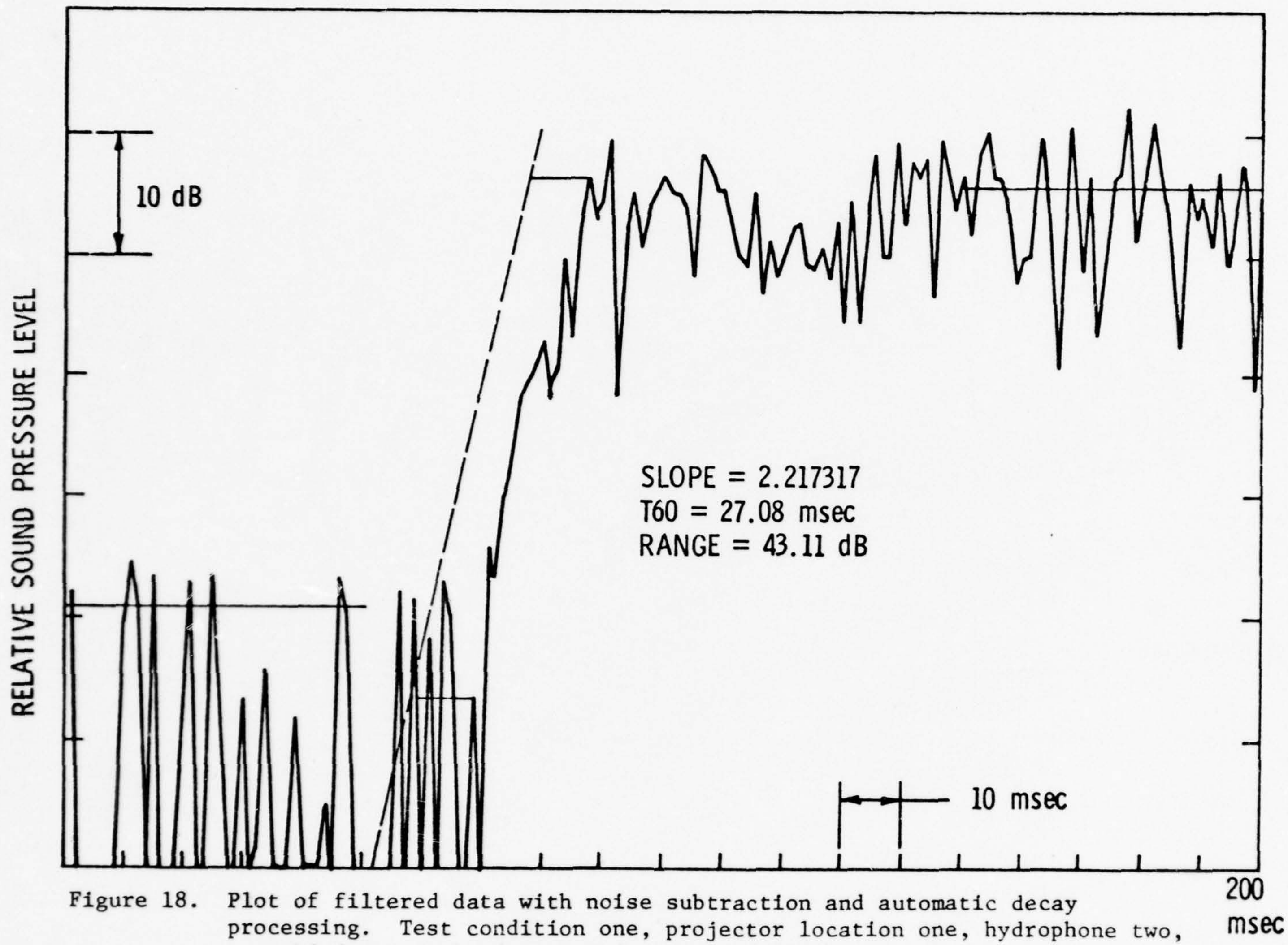

Figure 18. Plot of filtered data with noise subtraction and automatic decay processing. Test condition one, projector location one, hydrophone two, one-third octave band center frequency 4,000 Hz.

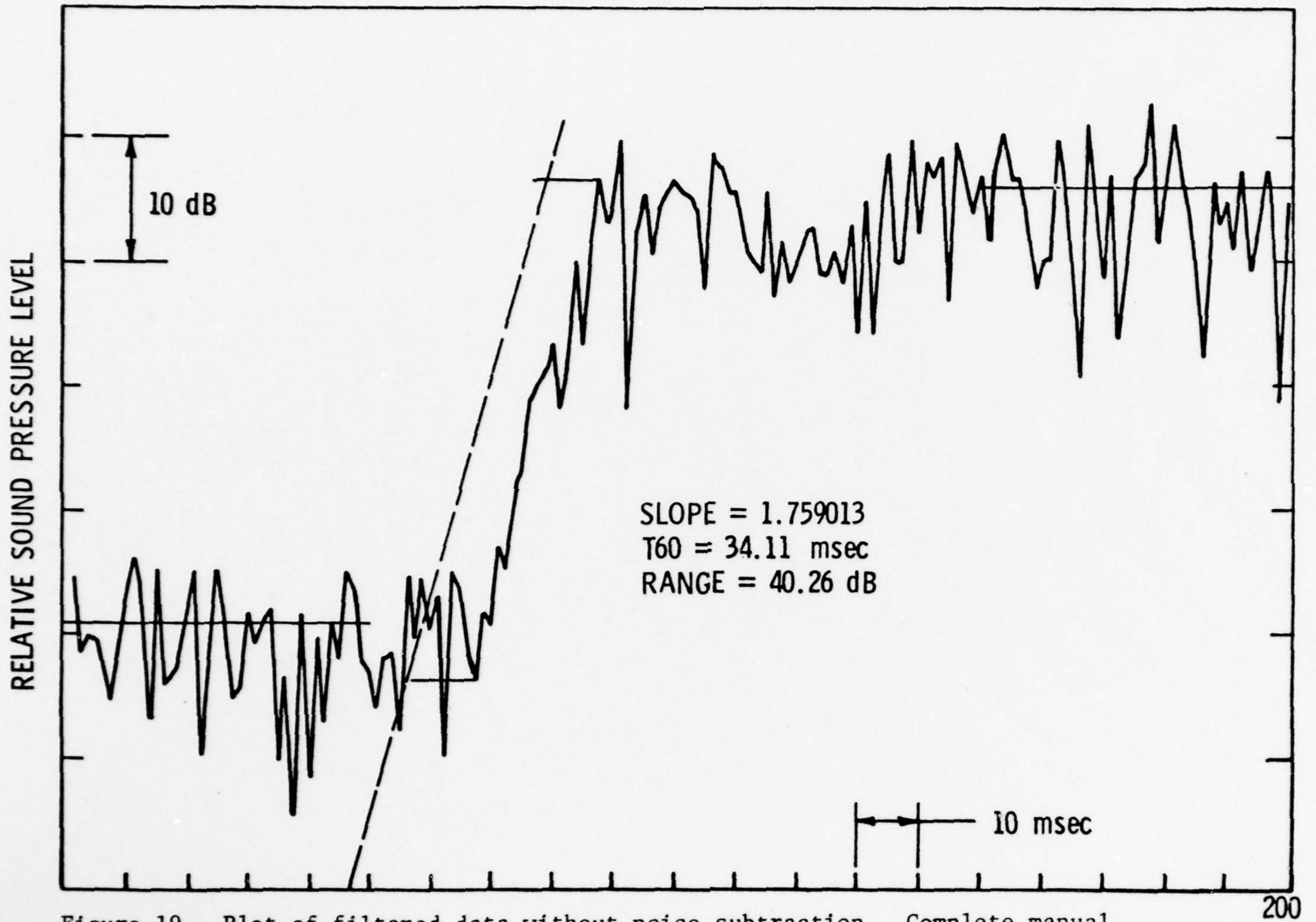

Figure 19. Plot of filtered data without noise subtraction. Complete manual selection of decay. Test condition one, projector location one, hydrophone two, one-third octave band center frequency 4,000 Hz.

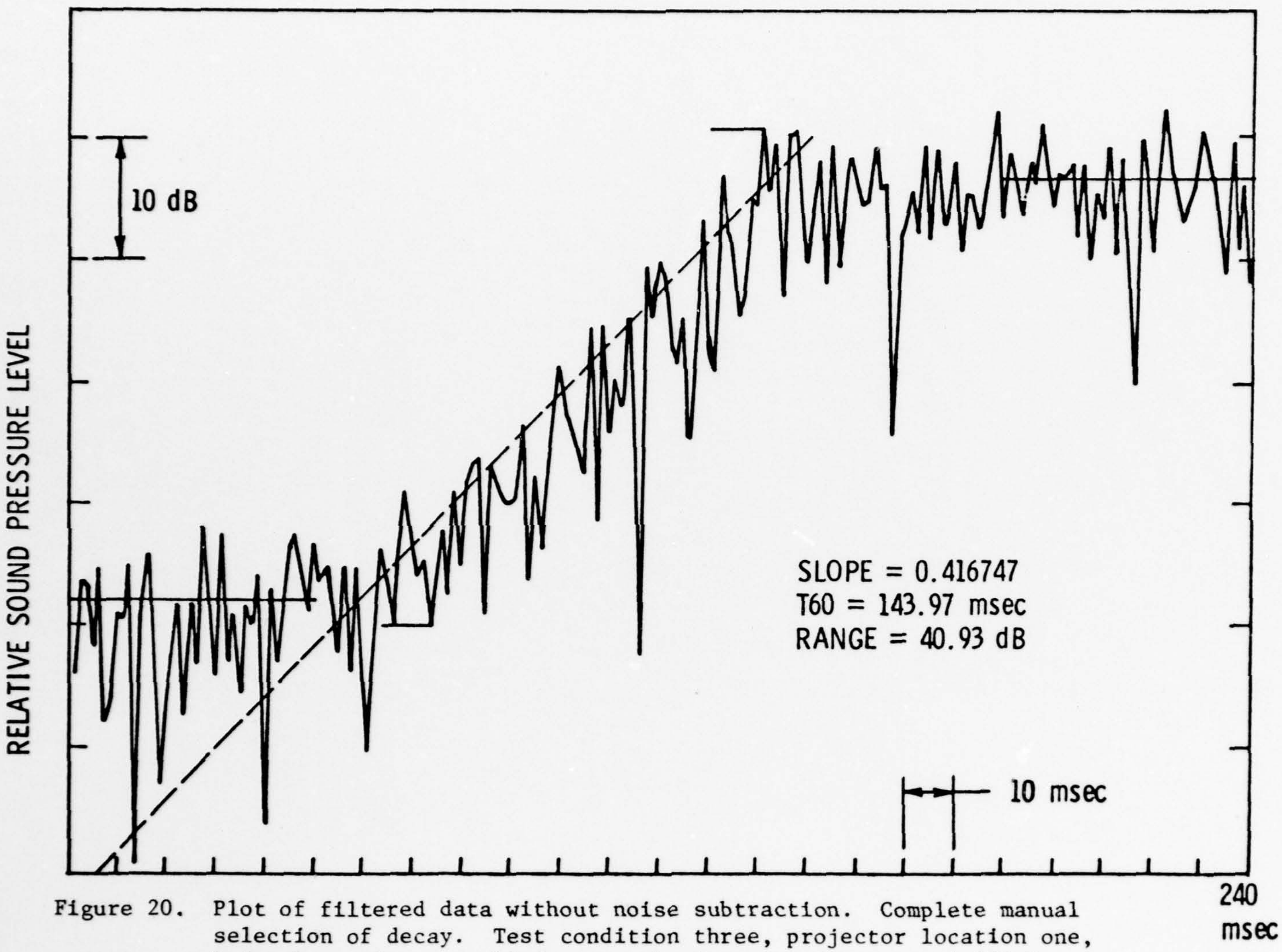

Figure 20. Plot of filtered data without noise subtraction. Complete manual selection of decay. Test condition three, projector location one, hydrophone six, one-third octave band center frequency 4,000 Hz.

this plot, the average background noise has not been subtracted from the data. As in the previous plot, the top and bottom of the decay, chosen by the operator in this case, are indicated in the plot along with the calculated slope and the average values of the background noise and excitation signal.

Figure 20 is a plot of a decay recorded under test condition three. As compared to the decays presented in the previous plots from test condition one, the decay in this plot is much longer with more fluctuations in the decaying sound level. It should be noted that the length of time represented in this plot is longer than in the previous plots which means that a direct, visual comparison of the slopes of the decays in these plots is not possible. It is possible, however, to see the extent of the change which resulted when the neoprene curtains were introduced into the MQL tank in test condition three.

4.2 Program Response Time Results

Before the full-scale analysis of the reverberation measurements was undertaken using program Reverb, the response time of the FFT filtering process was checked by using the program to analyze the transients which result when the signal from the random noise generator is shut off using the same toggle switch that was used for the reverberation measurements. These switch transients were recorded on tape and digitized using the same equipment that was used for the reverberation measurements. Table II shows the average reverberation times obtained by analyzing three separately recorded switch transients using each of the one-third octave bands approximated in the program.

TABLE II

SWITCH TRANSIENT AND FILTERED SWITCH TRANSIENT RESULTS

Switch Transient Results

Center Frequency: (in kHz)	1	1.25	1.6	2	2.5	3.15	4	5	6.3	8	10	12.5	16
Avg. Reverb. Time in ms	8.00	9.17	7.38	8.07	6.81	7.16	5.38	4.61	2.63	2.68	5.17	2.70	2.99
Std. Dev. in ms	0.66	0.73	1.26	0.95	0.69	1.28	0.52	1.27	1.08	1.29	0.95	0.59	0.22
Avg. Range in dB	18.90	17.76	24.31	23.34	22.25	21.43	24.03	21.35	28.42	32.85	29.96	30.22	30.10
Std. Dev. in dB	1.61	9.09	1.69	4.51	2.22	10.42	8.32	1.97	5.07	9.64	5.16	10.42	5.73

Filtered Switch Transient Results

Center Frequency: (in kHz)	0.50	0.63	0.80	1	1.25	1.6	2	2.5	3.15	4	5	6.3	8	10	12.5	16
Avg. Reverb. Time in ms	40.85	33.46	31.89	24.17	15.93	12.42	14.13	12.04	9.06	7.22	5.30	5.30	4.74	4.32	3.58	3.41
Std. Dev. in ms	4.63	2.62	1.96	2.62	1.89	4.20	0.12	0.30	0.35	1.10	1.58	0.01	0.23	1.14	1.26	0.71
Avg. Range in dB	32.75	34.78	30.60	34.32	37.00	37.44	42.64	33.51	38.94	35.77	33.77	36.35	34.82	29.40	31.75	28.30
Std. Dev. in dB	4.66	3.61	1.82	4.00	4.93	4.53	6.57	1.51	3.91	0.49	2.95	1.94	0.96	1.55	0.40	0.96

In a similar manner, the program was used to analyze the transients which occur when the switch transients are fed into the General Radio multifilter which had been used in the previously discussed analog analysis method. Since the bandwidth of the General Radio multifilter is exactly one-third octave wide, the FFT's selected for the analysis of the filtered switch transients had binwidths wider than one-third octave in accordance with standards for measuring reverberation time.[3] This was accomplished using the manual override capability of the FFT selection process in overlay one. Since wider binwidths are obtained by using smaller size FFT's, the time resolution obtained in the analysis of the filtered switch transients was better at low frequencies than that in the analysis of the switch transients or the reverberation data. As a result, the analysis of the filtered switch transients was performed down to a frequency one octave lower than the lowest frequency band possible for the switch transients or reverberation data.

Table II shows the average reverberation times of the filtered switch transients obtained by analyzing three separate filtered switch transient recordings for each frequency band. The results of both the switch transient analysis and the filtered switch transient analysis are represented in Figure 21 as the lower two curves on the graph.

By comparing the curves in Figure 21, it can be seen that the FFT filtering process, as implemented in program Reverb, responds faster to transients than does the General Radio multifilter, particularly at the lower frequencies. This tends to indicate that the problem of the slow response time of analog filters at low frequencies can be overcome using the technique of approximating

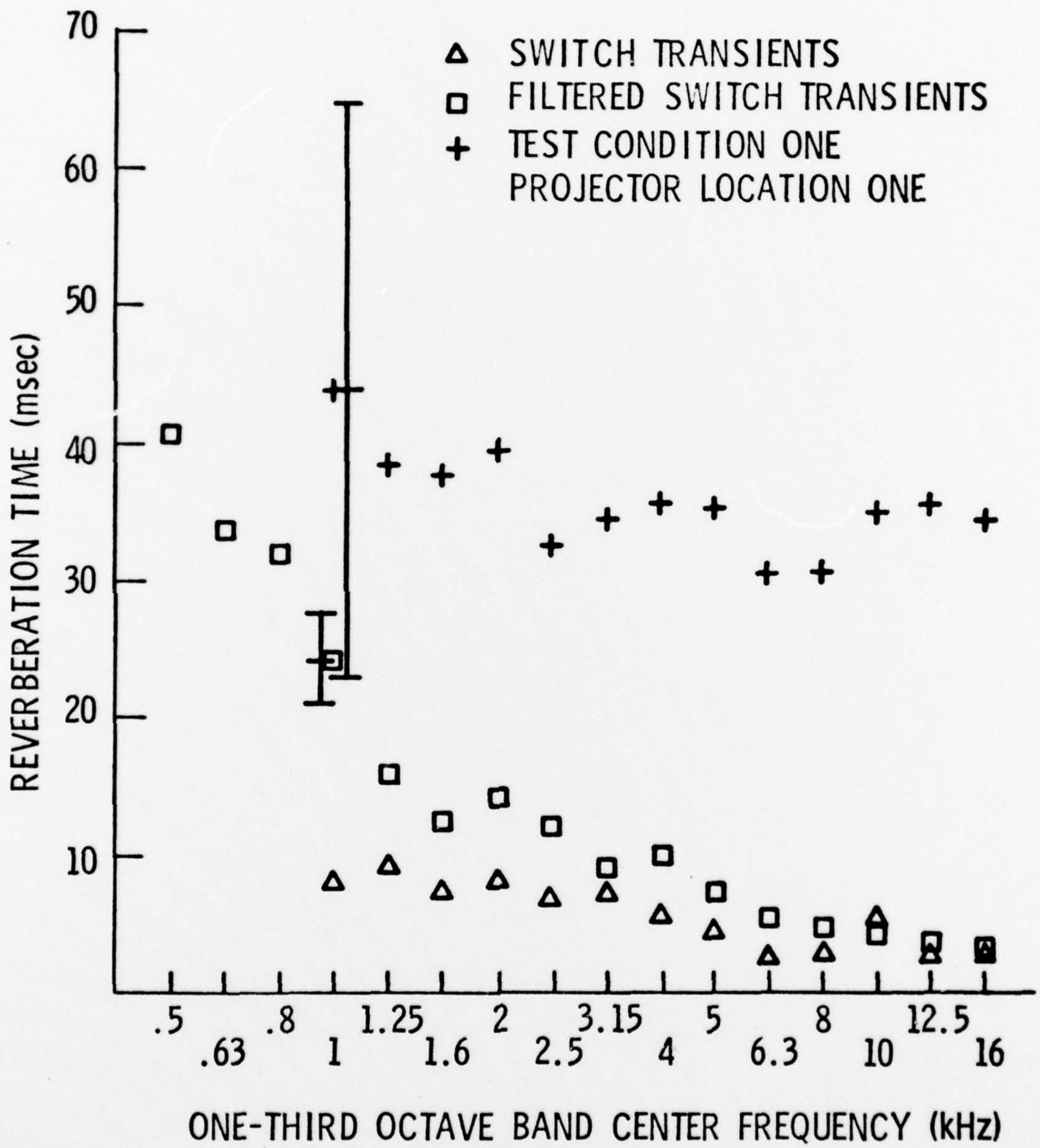

Figure 21. Graph of reverberation time as a function of one-third octave band center frequency for switch transients, filtered switch transients, and test condition one, projector location one.

one-third octave band filters using FFT's. The use of this technique is not without its difficulties, however. The execution time of the analysis program was about eight minutes for the frequencies not using the Hamming window method for increasing the time resolution of the plot. When the Hamming window technique was used, the execution time increased to almost twenty minutes. However, these long execution times are indications of the shortcomings of the computer system rather than the inefficiency of the program. A decrease in the execution time would occur if a faster computer were used with a magnetic disc system for storing the overlays rather than the present cassette system.

4.3 Reverberation Time Results

The results of the analysis of the data from test condition one, projector location one are given in Table III. The reverberation times given are the averages of the two data sets recorded at eight hydrophone positions for each frequency band. In addition, the average range over which the slope of the decay was determined is given along with the standard deviations of both the range and the reverberation time. The results are presented in Figure 21 as the top curve, and in Figure 22 as the bottom curve. The ranges of test condition one results and the filtered switch transient results at 1 kilohertz are represented by bars in Figure 21. It should be noted that the range of the filtered switch transient results overlaps the range of the test condition one results. This overlap illustrates that the response of analog filters can indeed interfere with the measurement of the reverberation time as was suspected in the analog and hydrid analysis methods.

TABLE III

REVERBERATION TIME RESULTS

Center Frequency: (in kHz)	1	1.25	1.6	2	2.5	3.15	4	5	6.3	8	10	12.5	16
Test Condition One, Projector Location One													
Avg. Reverb. Time in ms	43.87	38.22	37.71	39.39	32.58	34.57	35.61	35.26	30.43	30.53	35.05	35.62	34.22
Std. Dev. in ms	11.03	7.36	8.42	8.90	5.43	5.31	3.88	4.59	7.00	3.08	3.90	4.44	4.83
Avg. Range in dB	28.58	30.74	32.55	30.20	32.92	32.57	34.94	28.46	33.27	31.64	33.80	30.14	27.98
Std. Dev. in dB	8.28	5.06	4.74	3.27	5.58	5.45	8.62	4.32	7.45	6.80	6.86	3.96	3.31
Sab. Absorp. $S\alpha_{Sab}$ in m^2	33.74	38.73	39.25	37.58	45.43	42.82	41.57	41.98	48.64	48.48	42.23	41.55	43.25
Absorp. Coeff.	0.42	0.48	0.49	0.47	0.57	0.53	0.52	0.52	0.61	0.61	0.53	0.52	0.54
Test Condition Three, Projector Location One													
Avg. Reverb. Time in ms	108.81	117.42	112.93	113.46	119.71	121.46	121.14	116.48	117.31	113.37	114.57	116.17	114.92
Std. Dev. in ms	24.13	15.30	30.72	30.67	29.10	28.87	31.70	28.90	31.87	32.48	31.88	31.58	28.14
Avg. Range in dB	27.69	27.35	30.72	30.67	29.10	28.87	31.70	38.90	31.87	32.48	31.88	31.58	28.14
Std Dev. in dB	4.59	3.59	3.71	7.14	5.87	7.01	4.56	7.12	5.84	7.24	5.00	5.65	3.76
Sab. Absorp. $S\alpha_{Sab}$ in m^2	13.60	12.61	13.11	13.05	12.36	12.19	12.22	12.71	12.62	13.06	12.92	12.74	12.88
Absorp. Coeff.	0.17	0.16	0.16	0.16	0.15	0.15	0.15	0.16	0.16	0.16	0.16	0.16	0.16

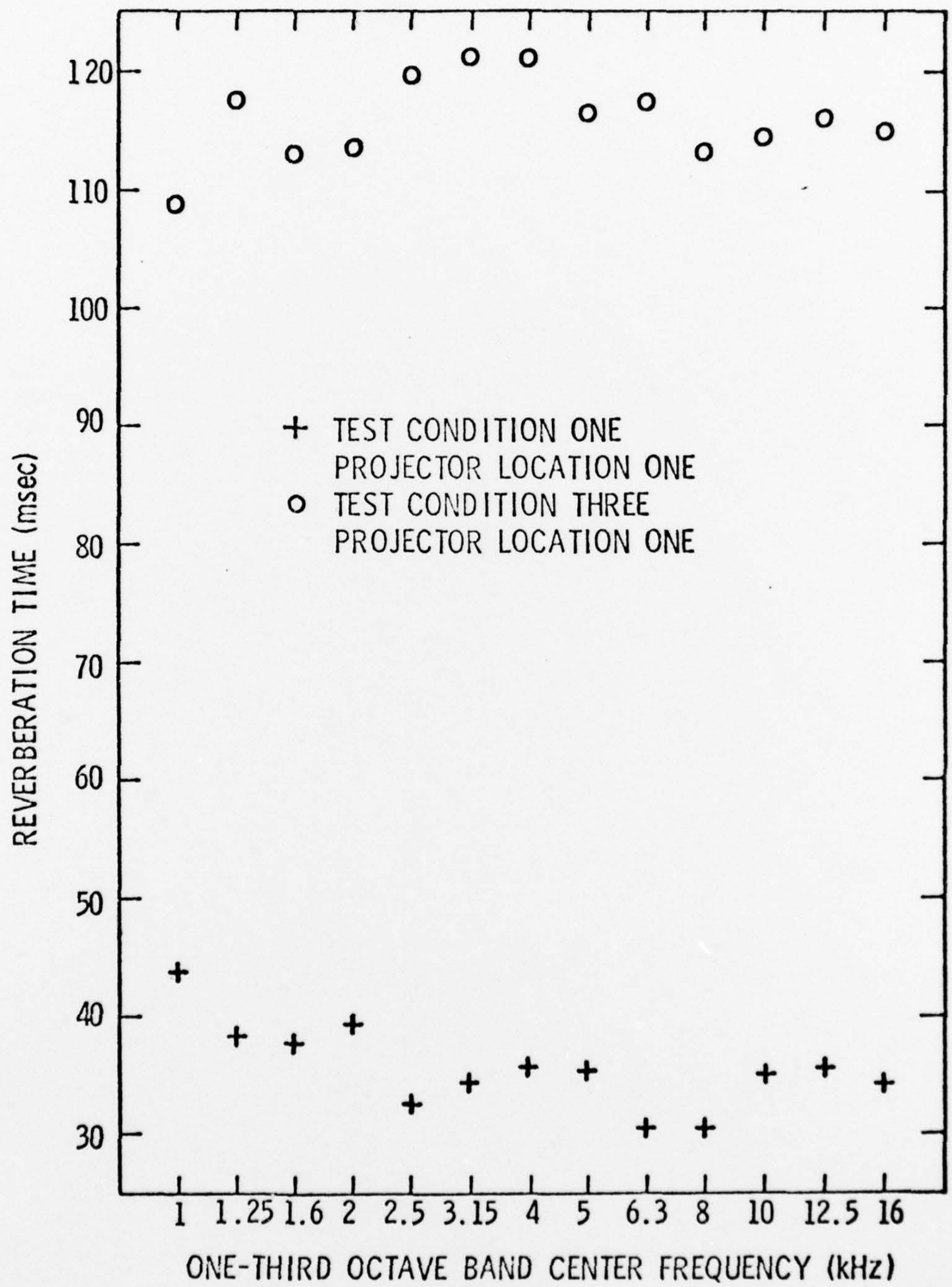

Figure 22. Graph of reverberation time as a function of one-third octave band center frequency for test condition one, projector location one, and test condition three, projector location one.

In addition, the shape of the curve of the filtered switch transient results tends to confirm the inverse relationship between the apparent reverberation time and the bandwidth of the filter. Since the bandwidth of a one-third octave filter is a constant percentage of the center frequency of the filter, the apparent reverberation time will double each time the center frequency of the filter is lowered by an octave. This relationship is similar to the speed reduction relationship for an analog tape recorder. For each reduction in the speed of the tape by one-half, the frequency of the signal is lowered by one octave, and the reverberation time of the recorded decay is doubled. Considering these two relationships, it can be seen that reducing the playback speed of an analog tape recorder offers no advantage in overcoming the problem of the limited response of analog filters.

The results of test condition three, projector location one are also included in Table III and depicted as the top curve in Figure 22. The results of test condition three show increases in the average reverberation times of about 2.5 to 3.5 times the results of test condition one. Such large increases show that by introducing the neoprene curtains into the MQL tank, the absorption in the tank was substantially reduced. Besides the longer reverberation times, the effect of the neoprene curtains was also observed in the lower gain settings that were recorded when the measurements were being taken. Since the neoprene curtains reflect sound better than the bare concrete walls, more of the sound energy radiated by the projector remained in the tank in the reverberant field when the neoprene curtains were in the tank than when they were not. The

hydrophones, responding to the increased sound pressure due to the larger contribution of the reverberant field, produced stronger signals which needed less amplification. As a result, when the neoprene curtains were in the tank, the amplifier gain settings necessary for the proper recording of the data were up to 20 decibels lower than the settings when the curtains were not in the tank.

The Sabine absorption ($S\alpha_{Sab}$) and the absorption coefficients (α_{Sab}), as determined by the relation

$$T_{60} = \frac{60\ V}{1.086\ c\ S\alpha_{Sab}} \quad ,$$

where V is the volume in m^3, and c is the speed of sound in m/s are also included in Table III. As can be seen from the Sabine absorption coefficients of test condition one, the MQL tank cannot be considered by air acoustics standards to be a reverberant chamber under test condition one due to the high absorption present as indicated by absorption coefficients much greater than 0.2 over the entire frequency range. The Sabine absorption coefficients of test condition three, however, are all below 0.2, indicating that the installation of the neoprene curtains in the tank resulted in a reduction of the absorption in the tank to values acceptable for a reverberation chamber by these same standards.

CHAPTER V

CONCLUSIONS

The methods employing analog filters to restrict the data to one-third octave bands were found to be unusable due to the limited response of the filters at low frequencies. To overcome this problem, a computer program was written to analyze the data using Fast Fourier Transforms to produce the desired filtering effect. The technique of using Fast Fourier Transforms as filters proved more effective for handling the decays of the reverberation measurements at low frequencies than the analog filters. Using this filtering technique along with algorithms for finding the decay and calculating the reverberation time from its slope, a full-scale computer analysis of the MQL reverberation measurements was started.

The results obtained when the neoprene curtains were not in the tank indicated that the concrete walls of the tank are too absorbent for the tank to be considered to be an underwater reverberant chamber based on air acoustics standards. However, with the neoprene curtains installed in the tank, the absorption is reduced enough for the MQL tank to be acceptable as a reverberant chamber by these same standards.

FOOTNOTES

1. W. K. Blake and L. J. Maga, J. Acoust. Soc. Am. 57, 380 (1975).
2. W. T. Chu, J. Acoust. Soc. Am. 63, 1444 (1978).
3. ANS S1.7-1970 (ASTM C 423-66) (R1972).

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